

Optranex 248 PSTN Gateway Field Trial Plan of Record

Optranex 248 PSTN Gateway

Field Trial Plan of Record

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Abstract:

This document provides a Field Trial Plan of Record for the Optranex 248 PSTN Gateway.

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The OpenSS7 Project <<http://www.openss7.org/>>

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Executive Overview

This document provides a *Field Trial Plan of Record* for the *Optranex 248 PSTN Gateway*. The initial and primary purposes of this equipment is to provide a scalable, carrier-grade platform for interconnecting VoIP backbone networks to the PSTN using SS7.

The OpenSS7 Project

The **OpenSS7 Project** is an open source software project that has developed many protocol components within the SS7, SIGTRAN, ISDN and VoIP protocol stacks. Intellectual property rights for the OpenSS7 Project are held by **OpenSS7 Corporation**. All OpenSS7 Project software is eventually licensed under the *GNU Affero General Public License Version 3*. OpenSS7 Corporation also provide commercial licensing of OpenSS7 Project software under terms less restrictive than the AGPLv3.

The OpenSS7 Carrier VoIP Switch

OpenSS7 can provide VoIP gateway capabilities in a high-performance, low-cost, small-footprint platform leveraging the *GNU/Linux* operating system distributions and tools, and utilizing low-cost commodity, or high-quality standardized hardware.

For details on platform applications, see [Chapter 2 \[Application Architecture\]](#), page 17 and [Chapter 4 \[Network Architecture\]](#), page 21.

Open Source Software

The OpenSS7 Project leverages the widespread use of GNU/Linux operation systems, distributions, and FSF tools such as ‘**autoconf**’ and *RPM*. For example, this document was formatted for PDF, HTML, info and plain text using the GNU *texinfo* system, ‘**autoconf**’, and the $\text{\TeX}$ formatting system.

The open source model avoids proprietary lock-in and permits in-house or outsourced development. All source code is available for use and modification by the end customer. All build tools, documentation and associated resources are generally available. The availability of the source code and complete documentation eases problem resolution and can offer upgrades and fixes even in advance of client problem reports.

For details on software solutions, see [Chapter 6 \[Protocol Architecture\]](#), page 37 and [Chapter 7 \[Software Architecture\]](#), page 93.

Commodity Hardware

By best utilizing commodity PC or standardized CompactPCI hardware, OpenSS7 makes available the highest performance platforms available on the market at back-to-school prices. When carrier-grade and large scale is not essential, 3GHz Pentium class servers in hardened rack mount chassis can be used at a fraction of the cost, and yet outperform, other solutions. Where carrier-grade is necessary, embedded Linux on standardized PICMG 2.16 NEBS compliant chassis make for a higher cost, but more reliable alternative.

For details on hardware solutions, see [Chapter 9 \[Platform Architecture\]](#), page 131 and [Chapter 8 \[Hardware Architecture\]](#), page 127.

Rapid Development

The OpenSS7 Project has already developed protocol components completing the SS7 and SIGTRAN signaling stacks including MTP Level 2 and Level 3, ISUP, SCCP, TCAP; and SCTP, M2PA, M2UA, M3UA, SUA and TUA. Development of a Carrier VoIP Switch to meet initial field requirements needs only the development of some intermediate and auxiliary modules.

For details on scheduling, see [Chapter 12 \[Logistics\]](#), page 137.

An Evolving Solution

The OpenSS7 Project is evolving to support more protocol stacks including ISDN and VoIP.

Conclusions

In summary, an OpenSS7 Carrier VoIP Switch an excellent application of the OpenSS7 SS7 and VoIP stacks and can be provided at a affordable price on short time-lines, while offering an evolution path for future deployment applications.

Brian Bidulock
The OpenSS7 Project

Preface

Document Information

Abstract

This document provides a *Field Trial Plan of Record* for the *Optranex 248 PSTN Gateway*.

Objective

The objective of this document is to provide a *Field Trial Plan of Record* for the deployment of the low cost, high-performance, *Optranex 248 PSTN Gateway* using OpenSS7 protocol components, software, and compatible systems and hardware.

Intent

The intent of this document is to act as a *Field Trial Plan of Record* for an project for an *Field Trial Plan of Record*. As a *Field Trial Plan of Record*, this document discusses components and systems which are not necessarily complete. **OpenSS7 Corporation** is under no obligation to provide any software, system or feature listed herein.

Audience

This document is intended for a technical audience. The reaser should be familiar with most most 3GPP, ETSI, ITU-T and ANSI, SS7, IMS, H.323, H.225, H.245, as well as IETF drafts and RFCs for RTP, SIP, SIP-T, MEGACO, MGCP, and SIGTRAN protocols.

Revisions

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Version Control

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ISO 9000 Compliance

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1 Introduction

This document provides a *Field Trial Plan of Record* for a platform to provide scalable *Optranex 248 PSTN Gateway* capabilities. The primary driver for the *Optranex 248 PSTN Gateway* is to provide a system capable of interconnecting a SIP VoIP backbone to the PSTN using SS7. This document provide a *Field Trial Plan of Record* for a trial system to provide this capability at a number of scale points.

The proposal utilizes, where possible, existing OpenSS7 protocol stack components and provides a development plan for components that are specific to the *Optranex 248 PSTN Gateway* initial requirements.

This document discusses the resulting software configuration that will be put in place on the production system, the platform configuration for the production system, and a network configuration for deployment. Also discussed is an overview of the project management logistics for successful completion over the course of this development project.

It is intended that the document be a “living” document, that is updated over the course of this development project.

1.1 The Optranex 248 PSTN Gateway

This project provides an *Optranex 248 PSTN Gateway* platform that provides interconnection to the PSTN using SS7 ISUP trunks and is capable of routing calls to and from the PSTN and a VoIP backbone network using *SIP-T*, *ITU-T Q.1912.5* and *3GPP TS 29.163* signaling and RTP transport.

1.2 Project Drivers

The lead purpose of the *Optranex 248 PSTN Gateway* is to provide PSTN interconnection to an existing VoIP backbone and deployment infrastructure that lacks same.

1.2.1 Project Objectives

The objectives that *Optranex Incorporated* wishes to achieve during the trial are as follows:

1. Validation of correct implementation of the *Optranex 248* switch hardware, firmware and software drivers. This will be achieved by exchanging channel data with the AT&T interconnection using aggregated DS-3 over SONET.
2. Validation of correct fail-over using 1:1 APS on the *Optranex 248* switch hardware, firmware and software drivers. This will be achieved by completing hardware fail-over between cards on the redundant *Optranex 248* switch platforms.
3. Demonstration of the maximum handling capacity of the *Optranex 248* switch. The *Optranex 248* switch is capable of connecting 2 x OC-48s for a total of 96 DS-3s (approximately 64,000 channels) in a redundant configuration. In simplex configuration, each server is capable of 96 DS-3s for a maximum simplex configuration of 192 DS-3s (approximately 128,000 channels). This will be achieved by loading the platforms to their maximum channel handling capacity and then performing voice quality assessment on a number of channels.
4. Demonstrate standard (RFC 4666) M3UA operation with the SS7 signaling network.
5. Demonstrate standard (ANSI T1.113-2000) ISUP operation with the SS7 signaling network.
6. Demonstrate standard (RFC 3621) SIP operation with the SIP network.
7. Demonstrate basic SIP to ISUP interworking operation according to SIP-T (RFC 3398), *ITU-T Q.1912.5* and *3GPP TS 29.163 Release 9*.

1.3 Scope

Because of early deployment drivers yet requirements for scale, the *Optranex 248 PSTN Gateway* platform is constructed using hardened PC telephony hardware in a NEBS 3/ETSI compliant chassis providing carrier-grade serviceability and reliability. Non-carrier-grade platforms utilizing commodity PC hardware for lower scale or lower reliability installations are possible.

1.3.1 Phases

The longer term project is broken into the following phases:

*Phase 1*¹ *The initial phase of the project is intended to provide the capabilities of the Optranex 248 PSTN Gateway for PSTN to VoIP backbone tandem for H.323.*

Phase 2 The second phase of the project adds tandem gateway capabilities for SIP-T.

Phase 3 The third phase of the project adds network service capabilities.

Phase 4 The fourth phase of the project adds softswitch capabilities.

Phase 5 The fifth phase of the project completes a VoIP (NGN) switch.

Although some reference is made to capabilities supporting other phases, *Phase 1* and *Phase 2* are the focus of this document.

1.3.2 Gates

Each phase of the project consists of seven gates. The seven gates are defined as follows:

Gate 0 — Concept

Gate 0 is passed when the initial concept has been elucidated and work is begun on a High-Level Design. This is an internal OpenSS7 gate.

Gate 1 — High Level Design

Gate 1 is passed when the high-level design has been reviewed to the satisfaction of the consumers of the project. This is an external review gate. OpenSS7 internally passes this gate once the High-Level Design has been published and work is begun on a detailed design.²

Gate 2 — Detailed Design

Gate 2 is passed when the detailed design has been reviewed to the satisfaction of the consumers of the project and the developers on the project. This is an external as well as an internal review gate. OpenSS7 passes this gate once the Detailed Design has been published and work base begin on development and implementation of the design.³ Passing this gate moves from the design stage to the development stage of the project.

¹ Although some reference is made to capabilities supporting other phases,

Phase 1 and *Phase 2* are the focus of this document.

² This document is a High-Level Design an Proposal document and it meets the internal requirements for passing Gate 1 of Phase 1 and Phase 2 of the VoIP GW project. An external review of this document by a Beta or Gamma client or sponsor is pending.

³ OpenSS7 requires a contractual commitment for purchase from a Beta or Gamma client, or funding from a Sponsor of the OpenSS7 Project, before this gate can be passed and development started.

Gate 3 — Development and Implementation

Gate 3 is passed when the software and systems development and implementation to the detailed design is complete and testing has begun. This is an internal review gate. OpenSS7 internally passes this gate when software is code complete and hardware has been installed for testing.

Gate 4 — System Test

Gate 4 is passed once the product implementation meets all internal ad hoc and formal conformance test suites and internal testing is complete. This is an internal review gate. OpenSS7 passes this gate internally once conformance testing is complete. Passing this gate moves from the development stage to the support stage of the project.

Gate 5 — Acceptance Test

Gate 5 is passed once the product implementation has passed external Gamma client acceptance testing. This is an external review gate. OpenSS7 passes this gate internally once participation in external acceptance testing is complete.

Gate 6 — Project Complete

Gate 6 is passed once all support obligations for the product implementation have been discharged. This is an internal review gate. OpenSS7 passes this gate once support agreements have terminated.

For more details on Gate scheduling for Phase 1 and Phase 2 of the VoIP GW project, see [Chapter 12 \[Logistics\]](#), page 137.

2 Application Architecture

The *Optranex 248 PSTN Gateway* is intended to provide high performance and high density PSTN to VoIP backbone gateway adding SS7 interconnection capability to an existing or back-end VoIP network.

2.1 Application Requirements

Application requirements have been broken into 5 phases using the timeboxing approach.

2.1.1 Phase 1 Requirements

Phase 1 requirements provide a VoIP gateway capability that will connect an existing H.323 VoIP network to the PSTN using SS7 ISUP trunks.

- Integral Media Gateway for RTP/RTCP to G.703/G.704 G.711 A- and mu-law.
- Integral Signaling Gateway for SS7 ISUP and SS7 TCAP.
- Local Number Portability.
- Lawful Intercept.
- Integral Media Gateway Controller for H.323
- Integral Back End functions for H.323.
- Integral H.323 Gatekeeper.

2.1.2 Phase 2 Requirements

Phase 2 requirements provide SIP-T capabilities.

- Integral Media Gateway Controller for SIP-T.
- Integral Back End functions for SIP-T.
- Integral SIP redirect/proxy server.

2.1.3 Phase 3 Requirements

Phase 3 requirements provide network service capabilities.

- ENUM support.
- AAA support.
- EAP authentication.
- External PLMN authentication.
- GSM MSC/VLR capabilities.
- ETSI Tiphon 4 interworking.
- ISDN support on E interface.

2.1.4 Phase 4 Requirements

Phase 4 requirements expose internal interfaces to provide softswitch capabilities.

- J Interface (SIGTRAN) exposed.
- N Interface (MGCP, H.248/MEGACO) exposed.

2.1.5 Phase 5 Requirements

Phase 5 requirements complete full VoIP (NGN) switching.

- Support for directly attached H.323, SIP and MGCP IP Phone terminals.
- Full RAS support for internal H.323 gatekeeper and SIP proxy server.
- Full class 5 residential/business feature set.

3 Solution Architecture

Although the functions of *Media Gateway Controller*, *Media Gateway* and *Signaling Gateway* have been decomposed, and in the past these functional groups have been implemented on separate physical platforms, modern compute capacity and densities permit these functions to be integrated into a single physical platform without limitation. Open standard interfaces are utilized internal to the platform to permit a decomposed model to be split out and to permit 3GPP IMS CN compatibility as well as ETSI Tiphon Version 4 and Multi-Services Forum Version 3 compatibility.

3.1 *Optranex 248 PSTN Gateway* Deployment

In light of the foregoing, the solution architecture takes the form of an integrated PSTN Gateway capable of providing a number of functional groups in the traditional models. The *Optranex 248 PSTN Gateway* integrates the following functional groups while still permitting standard interfaces to be exposed for maximum deployment flexibility:

- Integral *Media Gateway Controller* for SIP-T.
- Integral *Media Gateway* for RTP/RTCP to PDH G.703/704 G.711 conversion.
- Integral *Signaling Gateway* for SS7 ISUP, TCAP and ISDN signaling conversion.
- Integral *Redirect/Proxy* server for SIP-T.

3.2 Distinguishing Features

The primary distinguishing feature of the *Optranex 248* switch is extreme reduction of the following:

- capital expense (3 orders of magnitude);
- power consumption (2 orders of magnitude);
- footprint (2 orders of magnitude);
- closed and proprietary software (erradicated);
- cost per channel (3 orders of magnitude).

3.2.1 Capital Expense Reduction

The capital expense reduction from competing solutions (Telcobridges) is approximately \$500,000 (1 x OC-48 worth of OC-3 traffic), reduced to \$5,000 (1 x OC-48) for the *Optranex 248* switch. This is 2 orders of magnitude lower than competing solutions. Capital expense reduction from current popular solutions (Genband) is approximately \$2,500,000 (1 x OC-48 worth of traffic), reduced to \$5000 (1 x OC-48) for the *Optranex 248* switch. This is 3 orders of magnitude lower than current popular solutions.

3.2.2 Power Consupcion Reduction

The power consupcion reduction is from 50 kWatt for competing and popular solutions to less than 1 kWatt for the *Optranex 248* switch. Power consumption is 2 orders of magnitude lower than competing solutions.

3.2.3 Footprint Reduction

The footprint reduction is from 2 full 7' bays (48 U) to 2 x 1U x 20" deep chassis. This is 2 orders of magnitude reduction in footprint.

3.2.4 Closed and Proprietary Software Reduction

Whereas competing solutions provide some open-source software, the *Optranex 248* switch is completed implemented with open and open-source software.

3.2.5 Cost per Channel Reduction

Cost per channel reduction over approaches used by FreeSwitch, Asterisk, OpenPBX and others is from approximately \$4.64 to \$5.65 a channel, to \$0.008. This is approximately 3 orders of magnitude reduction in per-channel costs.¹

3.3 Call Flows

This section provides some illustrative application call flows:²

¹ See *High Density SONET/SDH MG Interface for VoIP Trunking*, B. Bidulock, April 24, 2009.

² This section is not intended as a *Detailed Design*, but provides illustration only for these *Application Notes*. Refer to RFC XXXX for detailed information.

4 Network Architecture

The network architecture is illustrated in [Figure 4.1](#).

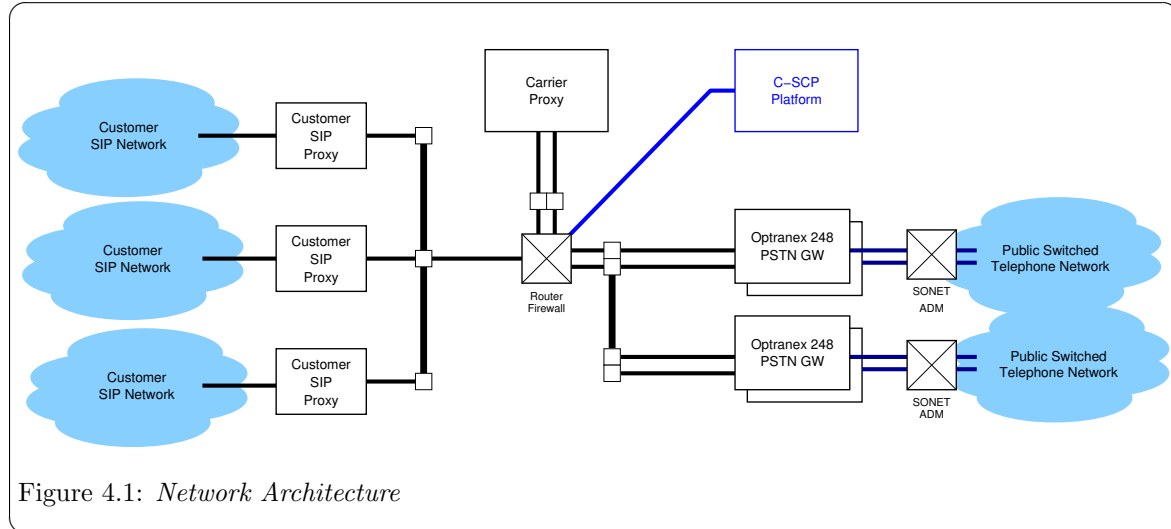


Figure 4.1: *Network Architecture*

The network architecture consists of the following major components:

- *Customer SIP Network* — A customer SIP network containing SIP UAs for originating and terminating calls. Each UA is associated with at least one NANP telephone number assigned to the customer SIP network.
- *Customer SIP Proxy* — An inbound and outbound SIP proxy used to pass calls to a Voice Peering Proxy (perhaps coexistent with the customer SIP proxy) and a PSTN GW.
- *Voice Peering Proxy* — A voice peering fabric proxy based on Carrier ENUM capable of determining whether outbound calls are destined for a Voice Peering fabric or the PSTN GW.

The purpose of a voice peering SIP proxy is to perform Carrier ENUM for the purposes of routing customer calls to a voice peering fabric or network for PSTN bypass. As there is no initial requirement for voice peering, this proxy is not shown in [Figure 4.1](#). Nevertheless, this proxy might be coexistent with the customer SIP proxy or the carrier proxy.

- *Carrier Proxy* — A telephony routing SIP proxy capable of Carrier ENUM or interaction with the C-SCP. This proxy might be coexistent with the C-SCP or an *Optranex 248* gateway switch.

The purpose of the carrier SIP proxy is to perform analysis and routing in conjunction with the C-SCP platform when necessary and determine to which of a number of *Optranex 248* gateway switches to route the call. Traffic estimates for 200 DS3sa requires the use of at least two *Optranex 248* gateway switches.

- *C-SCP Platform* — The *Converged-SCP* platform capable of number portability dips and the interworking with the carrier proxy as described in this document.
- *Router/Firewall* — An Internet Protocol Router/Firewall for the purpose of IP connectivity between the elements in the IP domain.
- *Optranex 248 Gateway Switch* — The *Optranex 248* gateway switch(es) connecting the IP and CS domains.

- *SONET ADM* — SONET Add-Drop Multiplexer(s) providing SONET connectivity between each *Optranex 248* switch and the PSTN.
- *Public Switched Telephone Network* — The PSTN consisting of Lata and Carrier access tandems for outbound and egress traffic.

In [Figure 4.1](#), the *Optranex 248* switch acts as a set of implicitly registered telephone subscriber User Agents towards the customer SIP proxies, and as an originating, terminating or tandem local switch to the PSTN.

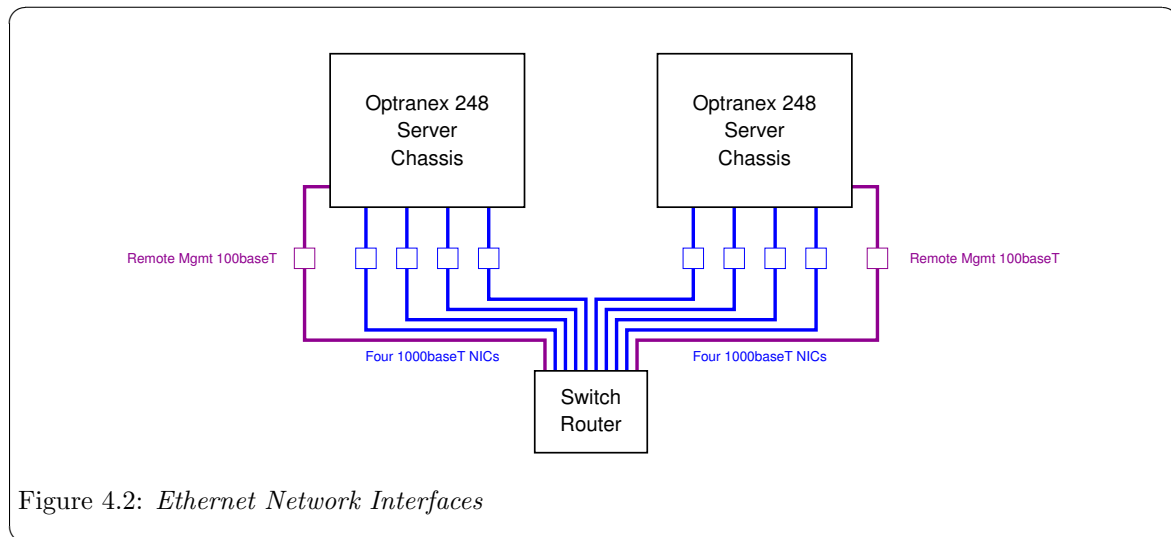
4.1 Ethernet Connections

As illustrated in [Figure 4.2](#), Ethernet connectivity of the *Optranex 248* switch will consist of four separate 1000BaseT Ethernet interfaces per server chassis for a total of eight separate 1000BaseT connections. These 1000baseT Network Interface connections are RJ-45 connectors that should be interconnected using Category 6 cable.¹

These 1000BaseT physical network interfaces will be used to support a number of separate subnetworks impressed on VLANs: one VLAN dedicated to each subnetwork.

In addition to the 1000BaseT ports described above, each server chassis has a single 100baseT port dedicated to an on-board Remote Management Module (RMM). This is also illustrated in [Figure 4.2](#). The 100baseT network interface connections are RJ-45 connectors that should be interconnected using Category 5 cable.²

These 100BaseT physical network interfaces do not require VLANs and may, instead, be connected on a dedicated non-VLAN subnetwork.



IEEE 802.1q prioritization will be used for prioritizing traffic on the various VLANs. The priority of the various VLANs from lowest to highest are as follows:

1. public Internet traffic;
2. test-bed RTP traffic;

¹ Shielded twisted pair (STP) is required for NEBS-3 compliance and the server chassis ports are designed to work with STP cable; however, the servers will operate with unshielded twisted pair (UTP) cable.

² Again, STP is required for NEBS-3 compliants but UTP will function.

3. test-bed SIP traffic;
4. test-bed *SIGTRAN M3UA* traffic;
5. customer RTP traffic;
6. customer SIP traffic;
7. *SIGTRAN M3UA* traffic;
8. management traffic.

IP Address Requirements

The IP address requirements for the *Optranex 248* are as follows:

- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN1 for public Internet access. These IP addresses must be able to address the public internet for software download and installation. These IP addresses can operate through a NAT to reach the public internet. Connection to the internet can be made intermittent or temporary for the purposes of security.
- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN2 for test-bed RTP traffic. These IP addresses need only address each other.
- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN3 for test-bed SIP traffic. These IP addresses need only address each other.
- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN4 for test-bed M3UA traffic. These IP addresses need only address each other.
- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN5 for customer RTP traffic. These IP addresses must be directly addressable from customer SIP networks (proxies or SBCs) and must be able to directly address customer SIP networks (proxies or SBCs).
- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN6 for customer SIP traffic. These IP addresses must be directly addressable from the carrier proxy (C-SCP) and must be able to directly address the carrier proxy (C-SCP). (These IP addresses might also need to directly address (and be addressed by) customer proxies.)
- Eight IP addresses (one for each 1000baseT port on each server chassis) assigned to the sub-network on VLAN7 for *SIGTRAN M3UA* traffic. These IP addresses must be directly addressable to and from the *SIGTRAN* Signaling Gateways.
- Ten IP addresses (one for each 1000baseT port, and one for each 100baseT port, on each sever chassis) with the eight (8) 1000baseT ports assigned to the subnetwork on VLAN8, and the two (2) 100baseT ports connected on a non-VLAN segment (but to the same subnetwork as VLAN8), for management traffic. This VLAN is expected to carry DNS traffic, NTP traffic, SNMP traffic, Remote Management Traffic, and SSH access from remote locations. These IP addresses must be able to address DNS and NTP servers. Also, some form of remote SSH or other access to the Remote Management Modules and login to the server chassis is required.

I would prefer if these addresses were not statically defined, but were acquired using DHCP and populated in DNS through either static or dynamic-DNS assignment from DHCP.

Traffic Requirements

The traffic requirements for each VLAN is listed in [Table 4.1](#), and detailed in the sections that follow:

Traffic Class	Source	Destination	Test Operation (Gbps)	Normal Operation (Gbps)
public Internet	A+B Internet	Internet A+B	– 1.0	– –
test-bed RTP	test-bed A+B A B	A+B test-bed B A	1.3 1.3 7.53 7.53	– – – –
test-bed SIP	test-bed A+B A B	A+B test-bed B A	0.004 0.004 0.012 0.012	– – – –
test-bed M3UA	test-bed A+B A B	A+B test-bed B A	0.004 0.004 0.012 0.012	– – – –
customer RTP	customer A+B	A+B customer	0.458 0.458	0.458 0.458
customer SIP	customer A+B	A+B customer	0.001 0.001	0.001 0.001
SIGTRAN M3UA	A+B SG	SG A+B	0.001 0.001	0.001 0.001
management	NMS/DNS/NTP A+B+RMM	A+B+RMM NMS/DNS/NTP	0.001 0.001	0.001 0.001

Table 4.1: *Traffic Requirements*

RTP Traffic Calculations

Each G.711 circuit can pass 8000 octets per second or 64 kbps. With a 10 millisecond packetization on AVT RTP G.711 PCMU, the RTP payload is 80 octets. With a 30 millisecond packetization on AVT RTP G.711 PCMU, the RTP payload is 240 octets. Each RTP has the following overheads:

- 18 octets Ethernet packet overheads.
- 4 octets VLAN packet overheads.
- 20 octets IPv4 header overheads.
- 8 octets UDP header overheads.
- 12 octets RTP header overheads.
- 62 octets Total overheads.

The packet size for 10ms packets with 80 octet payload is 142 octets or 1,136 bits. 10ms packets are generated 100 times per second per direction per channel, or 113,600 bits per second per direction. The efficiency of the 10ms RTP packet is approximately 56%. A DS3 of 672 channels requires a bandwidth of 76.3392 Mbps (in each direction) of 10ms RTP G.711 packets.

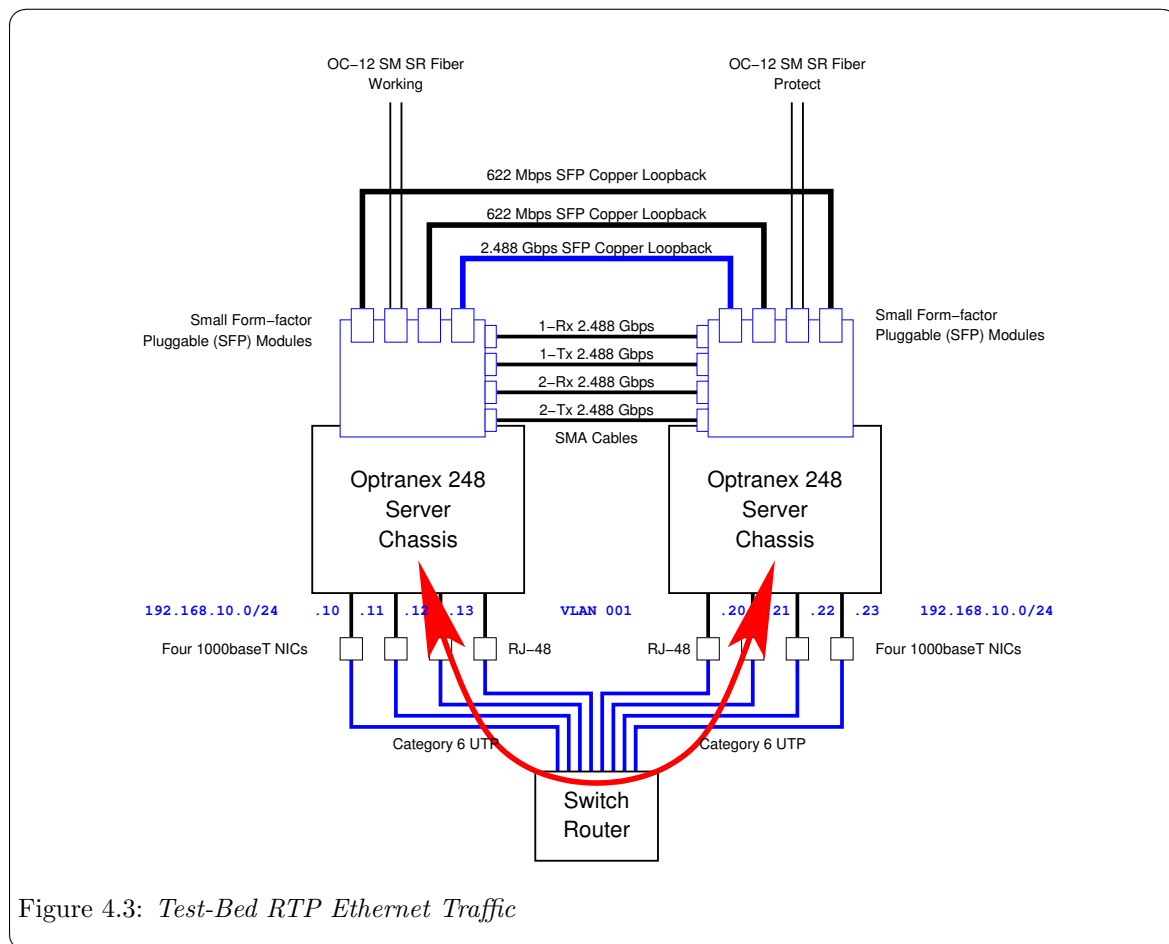
The packet size for 30ms packets with 240 octet payload is 302 octets or 2,416 bits. 30ms packets are generated 33-1/3 times per second per direction per channel, or 80,533 bits per second per direction. The efficiency of the RTP packet is approximately 80%. A DS3 of 672 channels requires a bandwidth of 54.1185 Mbps (in each direction) of 30ms RTP G.711 packets.

4.1.1 Internet Ethernet Traffic

The *Optranex 248* switch requires (temporary or permanent) connectivity to the public Internet for the purpose of downloading software loads and performing software upgrades. This is the lowest priority traffic. It is recommended that this traffic be performed using a VLAN dedicated to public Internet access. This VLAN should have a gateway that may be used as a default gateway and that provides NAT/Firewall access to the public Internet. It is anticipated that this connectivity to the public Internet will be used largely for installation and upgrade procedures only. As such, it can be temporary.

4.1.2 Test-Bed RTP Ethernet Traffic

The requirement for test-bed RTP Ethernet traffic is illustrated in [Figure 4.3](#).



For the trial, during performance and capacity testing, it will be necessary to pass a significant amount of test-bed RTP traffic between load generators and also intra-switch traffic between the two server chassis making up the *Optranex 248* switch. The primary purpose of test-bed RTP traffic is to accept traffic from load generators and voice quality assessment equipment.

It is recommended that Test-Bed RTP connectivity be performed using a VLAN dedicated to RTP traffic between the test-bed network and the *Optranex 248* switch. Separate IPv4 IP addresses for each of the network connections in each of the server chassis of the *Optranex 248* switch should be allocated on this dedicated VLAN subnetwork. All network interfaces of each *Optranex 248* service should be connected to this VLAN.

Four IP addresses per server chassis (total 8 IP addresses) on the test-bed RTP VLAN is required due to the number of RTP ports that are required. 64,512 channels requires 129,024 UDP ports, which, when spaced 2 apart (for RTCP), and consuming half of the available UDP port space, requires 8 IP addresses in a load-sharing configuration such as is used for performance testing. In an active-standby arrangement, each server would require 8 IP addresses of its own on this VLAN, and would also require 10G Ethernet connectivity.

For example, if VLAN 001 is used for Test-Bed RTP traffic, and Class C subnetwork 192.168.1.0/24 is associated with VLAN 001, the A-side of the *Optranex 248* platform could be assigned IP addresses 192.168.1.10, .11, .12 and .13 on VLAN 001, and the B-side of the *Optranex 248* switch could be assigned IP addresses 192.168.1.20, .21, .22 and .23. This configuration is illustrated in [Figure 4.3](#).

For maximum capacity performance testing, switches and routers on the Test-Bed RTP VLAN must be capable of passing 64,512 channels of G.711 RTP with 10 millisecond and 30 millisecond packetization. A 10 millisecond packetization consists of 80 octets of RTP AVT payload every 10 milliseconds for 64,512 channels full duplex or 2 x 6,451,200 packets per second. A 30 millisecond packetization consists of 240 octets of RTP AVT payload every 30 milliseconds for 64,512 channels full-duplex or 2 x 2,150,400 packets per second. Each RTP packet on the VLAN has 18 octets of Ethernet overhead, 4 octets of VLAN overhead, 20 octets of IPv4 overhead, 8 octets of UDP overhead, and 12 octets of RTP overhead, for a total of 62 octets of overhead. At 10 millisecond packetization that is 62 octets of overhead and 80 octets of payload at 2 x 6,451,200 packets per second or 14.65712 Gbps. At 30 millisecond packetization that is 62 octets of overhead and 240 octets of payload at 2 x 2,150,400 packets per second or 10.39073 Gbps. 4 x 1000baseT network connections are capable of 8 Gbps full duplex maximum: therefore, it will be possible to test 2 x 32,256 channels (2 x OC-48) or 2 x 7.32856 Gbps at 10ms packetization or 2 x 5.195365 Gbps at 30ms packetization during capacity testing. A single OC-48 (2.488 Mbps) SFP loop-back cable will be provided to permit 32,256 channels of loopback (2 x 64,512 channels) for capacity performance testing.

4.1.3 Test-Bed SIP Ethernet Traffic

For the trial, a test-bed SIP proxy and test-bed SIP softphones will also be connected on the same VLAN as the customer proxies.

For capacity performance testing, this VLAN must be capable of processing a maximum of 64,512 simultaneous calls, consisting of 3 minute average holding time. This consists of a maximum of 64,512 calls every 3 minutes, or 360 calls per second, which is on the order of 12 megabits per second of SIP signaling.

4.1.4 Test-Bed SIGTRAN M3UA Ethernet Traffic

It is recommended that test-bed *SIGTRAN M3UA* connection be performed using a VLAN dedicated to communications between the A- and B-side of the *Optranex 248* switch. Separate IPv4 IP

addresses for each of the server chassis of the *Optranex 248* switch should be allocated on this dedicated VLAN subnetwork. Router connectivity between the *Optranex 248* switch servers should be restricted to the MAC and IPv4 IP addresses of those servers. PoE or GRE may be used between the router and *Optranex 248* switch servers to simplify firewall designs for, and further isolate, this traffic.

4.1.4.1 Test-Bed SIGTRAN M3UA Network Configuration

SCTP Multihoming is (*RFC 4960*) is necessary to provide path and network interface redundancy for *SIGTRAN M3UA*. However, SCTP Multihoming does not function properly through a NAT. This problem can be side-stepped using SCTP-aware NAT (doesn't really exist yet in standard form) or by using LT2P, PoE, GRE or other point-to-point route encapsulation for establishing direct IP connections through the edge router.

As the IP addresses of the A- and B-side of the *Optranex 248* platform for test-bed *SIGTRAN M3UA* traffic may be wholly within a private IP network, route encapsulation is not necessary. (For the normal case, see [\[undefined\]](#), page [\[undefined\]](#).)

4.1.4.2 Test-Bed SIGTRAN M3UA AS Configuration

The *Optranex 248* switch servers will be configured as two M3UA IP Server Processes (IPSPs) serving separate Application Server(s). There will be one Application Server for each of the A- and B-side of the *Optranex 248* platform. Application Servers will be configured simplex (in standby or loadshare traffic modes).

Signaling Point Codes

The *Optranex 248* switch will require two small-network experimental SS7 signaling point codes for loop-back trunk configuration for capacity and performance testing. Four point codes will be initially selected from the highest available point code block in ANSI Signaling Point Code Block listing, *Table B3/T1.111.8*. The ANSI point code block assignments are listed in [Table 4.2](#).

ANSI Point Code Block Assignment	Platform Assignment
5-255-252	A-Side primary point code.
5-255-253	A-Side virtual point code.
5-255-254	B-Side primary point code.
5-255-255	B-Side virtual point code.

Table 4.2: *ANSI Point Code Block Assignments*

This point code will permit 4 trunk groups for testing to be configured:

Point Codes		Available Circuits	Assigned Circuits
A-Side	B-Side		
5-255-252	5-255-254	65536	32256
5-255-252	5-255-255	65536	8064
5-255-253	5-255-254	65536	8064
5-255-253	5-255-255	65536	8064
Total		262144	56448

Table 4.3: *Loop-Around Trunk Group Assignment*

These point codes will not be addressable from the public SS7 network and will only be addressable internally. Also, the trunk groups at the ends of which these signaling point codes are assigned will not terminate nor transit the PSTN, but will be configured as loop-around trunks between the A- and B-Side of the *Optranex 248* switch. See [\[undefined\]](#), page [\[undefined\]](#).

4.1.4.3 Test-Bed SIGTRAN M3UA Traffic Requirements

Sufficient bandwidth between the side of the *Optranex 248* gateway switch for *SIGTRAN M3UA* is required for proper test operation. For performance testing capacity, SONET connectivity consists of 2 OC-12 and 1 OC-48 of test traffic, for a total of 72 DS3s. With average 3-minute holding times, expected average call attempt traffic will be $(72 \times 672)/180$ or 268 calls per second. A typical ISUP call averages 5 messages with an average of 28 octets per message. Adding *SIGTRAN M3UA* overheads³ yields an average of 106 octets per message. At 268 calls per second, this totals a mere 227,264 bits per second.

Nevertheless, it is important that this traffic be given priority over less important flows (such as RTP traffic).

4.1.5 Test-Bed DNS Ethernet Traffic

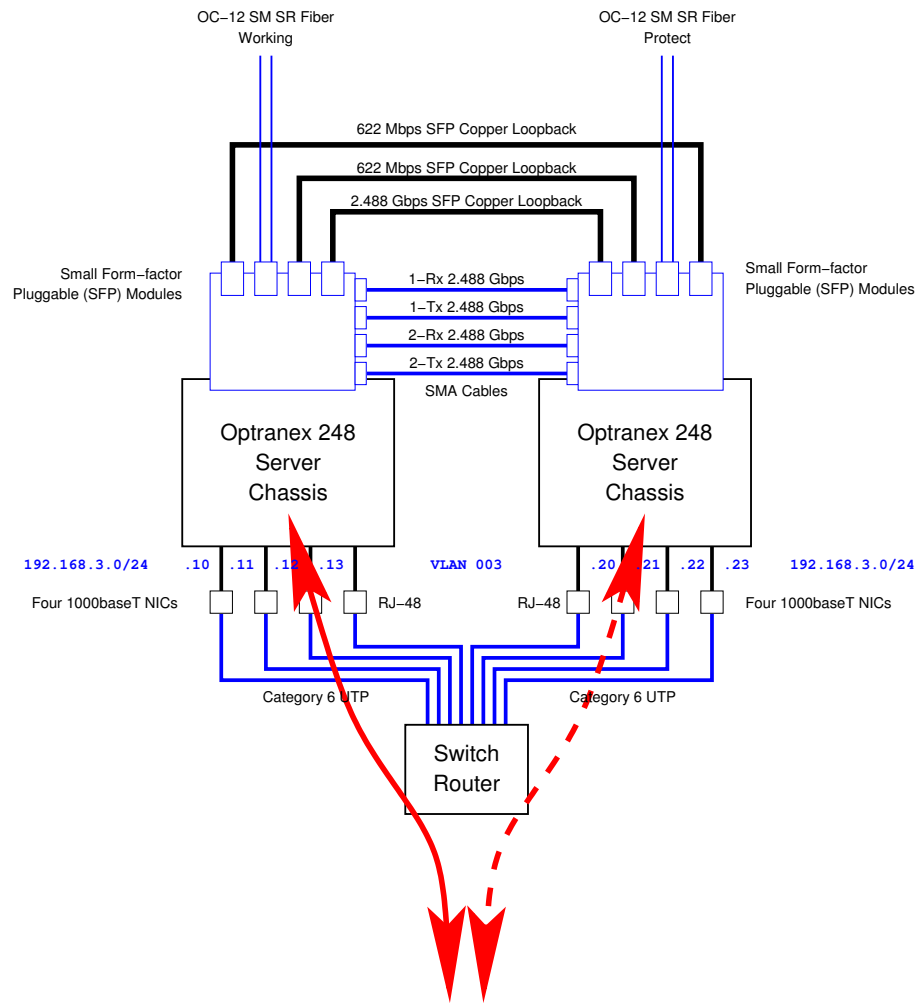
4.1.6 Customer RTP Ethernet Traffic

It is recommended that Customer RTP connectivity be performed using a VLAN dedicated to RTP traffic between the customer networks and the *Optranex 248* switch. Separate IPv4 IP addresses for each of the 1000baseT network interfaces on each of the server chassis⁴ of the *Optranex 248* switch should be allocated on this dedicated VLAN subnetwork. All four 1000baseT network interfaces on each *Optranex 248* server chassis should be attached to this VLAN subnetwork.

Although, for the trail, the amount of traffic is significantly lower than the maximum capacity of the *Optranex 248* switch, IP addresses should be allocated and network connections attached in accordance with the maximum capacity configuration of the switch. This avoids making disruptive modifications at a later date due to growth of the live platform.

³ 18 octets of Ethernet packet overhead, 4 octets of VLAN packet overhead, 20 octets of IPv4 header overhead, 16 octets of SCTP packet overhead, 8 octets of DATA chunk overhead, 12 octets of M3UA overhead.

⁴ If the optional 10Gig Ethernet connectivity is provided, this interface would require 4 separate IP address for the one interface.

Figure 4.4: *Customer RTP Ethernet Traffic*

Therefore, four IPv4 IP addresses per server chassis (total 8 IP addresses) on the customer RTP VLAN is required due to the number of RTP ports that are required in a maximum configuration. In a maximum configuration, 64,512 channels requires 129,024 UDP ports, which, when spaced 2 apart (for RTCP), and consuming half of the available UDP port space for a given IP address, requires 8 IPv4 IP addresses in a load-sharing configuration such as is used for performance testing. In an active-standby arrangement, the active server assumes all 8 IP addresses (using IP takeover if necessary) on this VLAN, and would also require 10GE connectivity.

For example, if VLAN 003 is used for Customer RTP traffic, and Class C subnetwork 192.168.3.0/24 is associated with VLAN 003, the A-side of the *Optranex* 248 platform could be assigned IPv4 IP addresses 192.168.3.10, .11, .12 and .13; the B-side, 192.168.3.20, .21, .22 and .23. This configuration is illustrated in [Figure 4.4](#).

For the trial, during normal operation (no capacity performance testing in effect), this VLAN must be capable of passing approximately 2016 channels of G.711 RTP PCMU with 10 millisecond and 30 millisecond packetization. A 10 millisecond packetization consists of 80 octets of RTP AVT

payload every 10 milliseconds for 2,016 channels full-duplex or 2 x 201,600 packets per second. A 30 millisecond packetization consists of 240 octets of RTP AVT payload every 30 milliseconds for 2,016 channels full-duplex or 2 x 67,200 packets per second. Each RTP packet on the VLAN has 18 octets of Ethernet overhead, 4 octets of VLAN overhead, 20 octets of IPv4 overhead, 8 octets of UDP overhead, and 12 octets of RTP overhead, for a total of 62 octets of overhead. At 10 millisecond packetization that is 62 octets of overhead and 80 octets of payload at 2 x 201,600 packets per second or 458.0353 Mbps. At 30 millisecond packetization that is 62 octets of overhead and 240 octets of payload at 2 x 67,200 packets per second or 324.7104 Mbps.⁵

4.1.7 Customer SIP Ethernet Traffic

It is recommended that Carrier Proxy SIP connectivity be performed using a VLAN dedicated to SIP signaling between the carrier proxy and the *Optranex 248* switch.⁶ Separate IPv4 IP addresses for each of the server chassis of the *Optranex 248* switch should be allocated on this dedicated VLAN subnetwork. Two 1000baseT network interfaces on each server chassis should be attached to this VLAN subnetwork. Network interfaces distinct from those used for *C-SCP* connection should be used.

For the trial, this VLAN must be capable of processing a maximum of approximately 2000 simultaneous channels of traffic, consisting of 3 minute average holding time calls. This consists of a maximum of approximately 2000 calls every three minutes or 12 calls per second, which is on the order of a megabit per second of traffic.⁷

4.1.8 SIGTRAN M3UA Ethernet Traffic

It is recommended that *SIGTRAN M3UA* connection be performed using a VLAN dedicated to communications between the *Optranex 248* switch and the PSTN SS7 network. Separate IPv4 IP addresses for each of the server chassis of the *Optranex 248* should be allocated on this dedicated VLAN subnetwork. Router connectivity between the *Optranex 248* switch servers and the *SIGTRAN M3UA* signaling gateway(s) should be restricted to the MAC and IPv4 IP addresses of those servers. PoE or GRE may be used between the router and *Optranex 248* switch servers to simplify firewall designs for, and further isolate, this traffic.

4.1.8.1 SIGTRAN M3UA Network Configuration

SCTP Multihoming (*RFC 4960*) is necessary to provide path and network interface redundancy for *SIGTRAN M3UA*. However, SCTP Multihoming does not function properly through a NAT. This problem can be side-stepped using SCTP-aware NAT (doesn't really exist yet in a standard form) or by using L2TP, PoE, GRE or other point-to-point route encapsulation for establishing direct IP connections through the edge router.

If Signaling Gateway (SG) IP addresses are within a private network, direct addressing is possible. If the SG IP addresses are in a public network or VPN external to the private network to which the *Optranex 248* is attached, route encapsulation to allow the *Optranex 248* to assign direct IP addresses allocated from the external network.

An additional requirement for the trial is that all SCTP addresses must be IPv4 addresses.

⁵ Note that if the entire OC-12 were to be utilized during the trial, the number of channels could increase by a factor of four to 8064, giving 1.833 Gbps of RTP traffic at 10ms packetization or 1.3 Gbps of RTP traffic at 30ms packetization.

⁶ Note that currently the *Carrier Proxy* is envisioned as being coexistent with the *C-SCP*.

⁷ Increasing the capacity at trial to the full OC-12 would not increase the customer SIP signaling bandwidth to significant levels.

4.1.8.2 SIGTRAN M3UA AS Configuration

The *Optranex 248* switch servers will be configured as two M3UA Application Server Processes (ASPs) serving the same Application Server(s). There will be one Application Server for each of the SS7 Signaling Point Codes assigned to the *Optranex 248* switch. Applications Servers should be configurable at the SG in either standby or loadshare traffic modes.

Signaling Point Codes

The *Optranex 248* switch will have at least one SS7 point code allocated across both A- and B-side server chassis.

4.1.8.3 SIGTRAN M3UA Traffic Requirements

Sufficient bandwidth between the *Optranex 248* gateway switch and the *SIGTRAN M3UA* signaling gateways is required for proper operation. For the trial, SONET connectivity consists of OC-12 SONET connections; however, only 3 DS3s worth of traffic is initially provided. With average 3-minute holding times, expected average call attempt traffic will be $(3 \times 672)/180$ or 12 calls per second. A typical ISUP call averages 5 messages with an average of 28 octets per message. Adding *SIGTRAN M3UA* overheads⁸ yields an average of 106 octets per message. At 12 calls per second, this totals a mere 10,176 bits per second.

Nevertheless, it is important that this traffic be given priority over less important flows (such as RTP traffic).

4.1.9 DNS Ethernet Traffic

4.1.10 Management Ethernet Traffic

It is recommended that management connectivity be performed using a VLAN dedicated to communications between management stations and the *Optranex 248* switch. Separate IPv4 IP address for each server chassis of the *Optranex 248* should be allocated on this dedicated VLAN subnetwork. All four 1000baseT network interfaces on each server chassis should be attached to this VLAN subnetwork. Separate IPv4 IP addresses for integrated management of the servers and management of the application should be provided. Management traffic is expected to be insignificant in terms of bandwidth.

4.1.10.1 DNS Ethernet Traffic

The *Optranex 248* requires access to DNS for both internal and external hosts to be able to resolve domain names.

4.1.10.2 NTP Ethernet Traffic

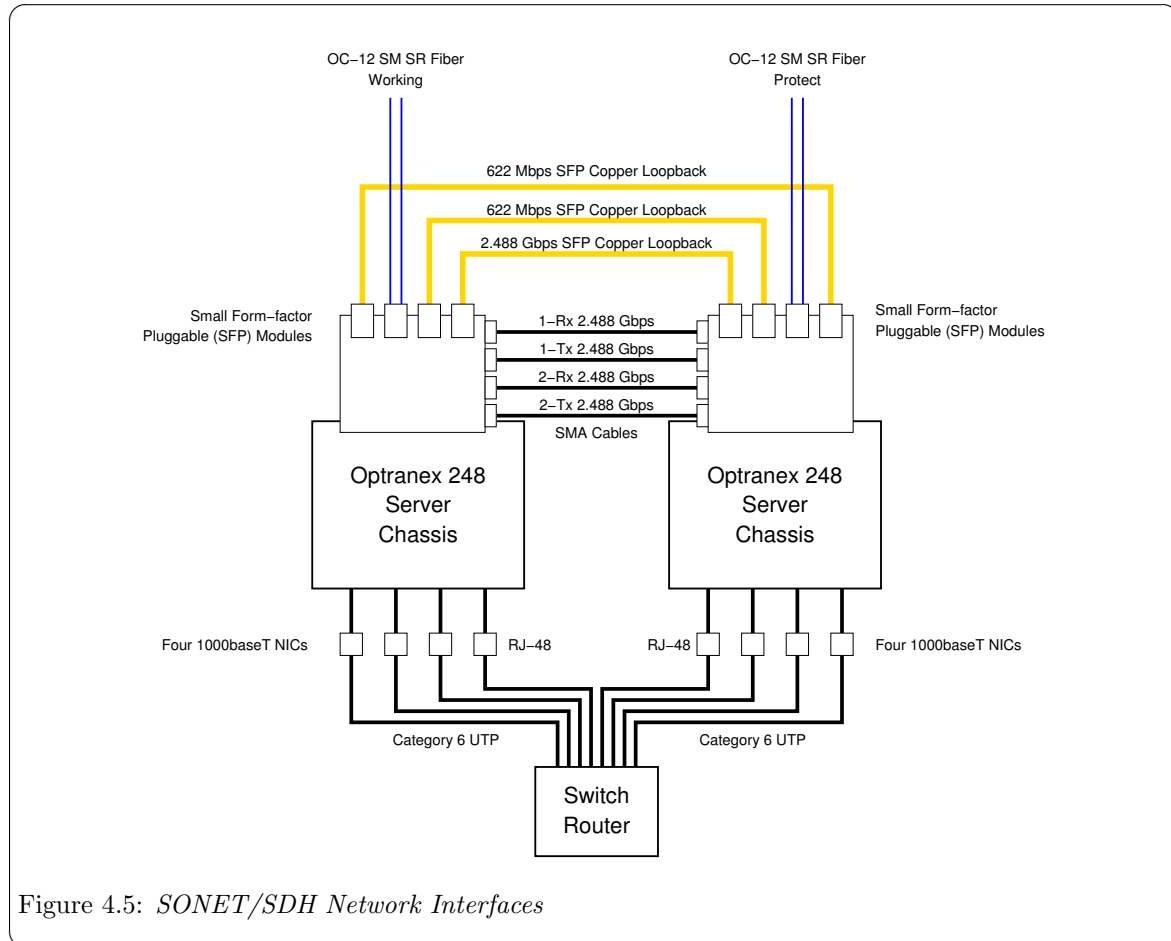
The *Optranex 248* requires a permanently available *NTP (Network Time Protocol)* source to synchronize the time-of-day clocks between the platforms and to the network. Should the NTP source be temporarily unavailable, the A- and B-sides of the platform will attempt to synchronize to each other with the A-side as master and the B-side as slave. A network NTP source can be provided using any (or all) of the standard methods: unicast request/response, multicast source, broadcast source. It is preferred that all methods operate on the Management VLAN.

Management stations and monitoring devices should be synchronized to the same time source as the *Optranex 248* platform.

⁸ 18 octets of Ethernet packet overhead, 4 octets of VLAN packet overhead, 20 octets of IPv4 header overhead, 16 octets of SCTP packet overhead, 8 octets of DATA chunk overhead, 12 octets of M3UA overhead.

4.2 SONET/SDH Connections

The SONET/SDH network connections are illustrated at the top of [Figure 4.5](#).



There are two primary SONET/SDH connections:

1. One PSTN Aggregated DS3 over SONET OC-12 connection to the PSTN for carrying live customer traffic and PSTN test traffic during the trial.
2. One copper SFP loopback connection for the purpose of capacity performance testing.

4.2.1 PSTN Aggregated DS3 over SONET

The primary traffic-bearing SONET/SDH connection for the trial consists of an OC-12 connection which will initially connect 3 aggregated DS3s (a maximum of 2016 channels). These trunks will be organized into trunk groups and assigned a number that will also be assigned (for trunk groups containing outgoing and two-way trunk members) in the *C-SCP* platform. It is recommended that the OC-12 connection to the interconnect ADM be completed as a 1:1 APS connection where the working OC-12 is connected to one of the two server chassis in the *Optranex 248* switch, and the protect OC-12 is connected to the other of the two server chassis in the *Optranex 248* switch.

4.2.1.1 Synchronization

The *Optranex 248* card contains a stratum 3 clock with accuracy for OC-192 operation. It is expected that aggregated DS3's are network synchronized to the SONET network or to a stratum 3 clock. If this is not the case, positive and negative pointer adjustment compensation on the Synchronous Payload Envelope for each aggregated DS-3 will be required.¹ The system side of the OC-48 framer on the *Optranex 248* card is frame-locked to the line side. Line-side timing can be derived from the SONET network received line signal or can be taken from the on-board stratum 3 clock.

When not connected to the SONET network, and for performance or capacity testing only, the A-side of the *Optranex 248* chassis can be configured to take its synchronization from the local stratum 3 clock; and the B-side of the *Optranex 248* chassis can be configured to take its timing from the receive SONET line.

When connected to the SONET network, both the A- and B-sides of the *Optranex 248* gateway switch should be configured to take their timing directly from the SONET receive line, with a fallback to the local stratum 3 clock (and a possible further fall back to DPLL hold).

4.2.2 Test-Bed Copper OC-48 Loop Back

For the purpose of capacity and performance testing, an additional SONET/SDH connection will be simulated using an SFP copper loopback cable capable of 2.488 Gbps operation. One end of this cable is attached to the A-side of the *Optranex 248* platform; the other, to the B-side. This connection will be capable of passing 32,256 channels between the A- and B-sides of the *Optranex 248* for capacity and performance testing.

4.2.3 Trunk Group Configuration

4.2.3.1 Controlling Exchange

It must be identified whether the *Optranex 248* or the remote exchange is the controlling exchange on all two-way trunks.

4.2.3.2 Circuit Management Groups

Circuit management groups will need to be identified. It needs to be specified whether circuit groups align with T1 circuit groups embedded in the aggregated DS3 or, if multiplexing or transmuxing is performed between the aggregated DS3 and the remote switches, the circuit group assignments.

4.2.3.3 Continuity Testing

It must be identified, for all trunks (whether one-way incoming, one-way outgoing, or two-way), whether automatic continuity testing is required and on what basis continuity testing is required (i.e. none, statistical, periodic). The period must be identified if periodic.

For calls terminating on the *Optranex 248*, for trial, I would prefer to skip the continuity check procedures. If this is permitted under interconnect agreement, I would prefer to release the call with cause value 41 (Temporary Failure) on all terminating calls with automatic continuity request in the IAM. This may increase the signaling load, particularly if (as AT&T does sometimes) continuity testing is performed on **every** call. If interconnect agreements require, the *Optranex 248* will be configured for proper installation of loopback on incoming automatic continuity request calls, and for automatic continuity rechecks. The requirements for the interconnect agreement need to be identified.

¹ I would prefer to avoid this unless absolutely necessary.

For calls originating from the *Optranex 248* (and dual-seizure), for trial, I would prefer to skip the continuity check procedures. First preference is to suppress continuity check for all outgoing circuits. As a second preference, should interconnect agreements require continuity checks on outgoing calls, again only for trial, I would prefer to blindly indicate successful COT when the procedures are invoked. The purpose of this is to initially avoid the need to implement the required digital filtering for continuity tone positive and negative detection. Because generation of COT tone is a less onerous requirement, COT tone can be generated but detection blindly accepted. For trial it needs to be identified as to whether full continuity testing

For outgoing calls I would be prefer to skip per-call continuity testing, for trial, and only configure for management initiated continuity tests for circuit turn-up.

If continuity testing is required periodically on outgoing calls under interconnect agreement, I would prefer to blindly indicate successful continuity test.

- release the call with cause value 41 (Temporary Failure);
- install a loopback (vs. hi-lo);
- allow the test to fail.

Also, for continuity tests on calls terminating to the *Optranex 248*, I would prefer, for the purposes of trial, a loopback arrangement.

4.2.4 Test Lines

For the purpose of test calls providing the capability of voice path quality assessment, a number of test lines will be necessary. Test line terminations will be required on, or via, interconnect switches on the PSTN. Test line terminations will also be provided on the *Optranex 248* switch. Test lines include

100	balance and noise test termination (type 100 test line) (balanced-termination)
101	test board communication (type 101 test line)
102	milliwatt test (type 102 test line) (milliwatt)
103	supervisory and signaling test circuit (type 103 test line) (not with SS7)
104	transmission measuring and noise checking (type 104 test line) (silent-termination)
105	automatic transmission measuring test (type 105 test line)
106	spare
107	data transmission test (type 107 test line)
108	digital circuit loopback test (type 108 test line) (loopback)
109	echo canceller test (type 109 test line)

5 Reference Architecture

There are several reference architectures to which the solution architecture applies, as follows:

1. IETF SIP Architecture
2. Multiservices Switching Forum Release 1, 2 and 3 Architectures
3. ETSI Tiphon Release 4 Architecture
4. ITU-T TR.45 Architecture
5. 3GPP Release 9 IMS Architecture

Of these, by far the most important is the *3GPP Release 9 IMS Architecture* as described in *3GPP TS 23.002 V9.2.0 (2009-12)*: the *3GPP Release 9 IMS Architecture* encompasses portions of the *ITU-T TR.45 Architecture* as it subsumes *ITU-T Recommendation Q.1912.5*; the *ETSI TIPHON Release 4 Architecture* has fallen asside for the *3GPP Release 9 IMS Architecture*; the *Multiservices Switching Form Release 3 Architecture* defers to the *3GPP Release 9 IMS Architecture* for all of the non-legacy components; and, the *IETF SIP Architecture* is antiquated for telephony.

The *3GPP Release 9 IMS Architecture*¹ consists of four or five elements of interest, and about as many interfaces of interest, in the current discussions:

<i>MRFC</i>	The <i>MRFC</i> is the <i>Media Resource Function Controller</i> . This function in the solution architecture is provided primarily by the <i>Optranex 248</i> switch.
<i>MRFP</i>	The <i>MRFP</i> is the <i>Media Resource Function Processor</i> . This function in the solution architecture is provided primarily by the <i>Optranex 248</i> switch.
<i>MGCF</i>	The <i>MGCF</i> is the <i>Media Gateway Controller Function</i> . This function in the solution architecture is provided primarily by the <i>Optranex 248</i> switch.
<i>MGF</i>	The <i>MGF</i> is the <i>Media Gateway Function</i> . This function in the solution architecture is provided primarily by the <i>Optranex 248</i> switch.
<i>BGCF</i>	The <i>BGCF</i> is the <i>Breakout Gateway Controller Function</i> . This function in the solution architecture is provided primarily by the customer proxy but partially by the <i>C-SCP</i> platform.
<i>I-CSCF</i>	The <i>I-CSCF</i> is the <i>Interrogating Call Services Control Function</i> . This function in the solution architecture is provided primarily by the customer proxy but partially by the <i>C-SCP</i> platform.
<i>S-CSCF</i>	The <i>S-CSCF</i> is the <i>Serving Call Services Control Function</i> . This function in the solution architecture is provided by the customer proxy.

Mn Interface (MGCF to MGW)

The *Mn* reference point describes the interfaces between the *MGCF* and *IMS-MGW* in the *IMS*. It has the following properties:

- full compliance with the H.248 standard functions for IMS—PSTN/PLMN interworking;
- flexible connection handling that allows support of different call models and different media processing pruposes not restricted to H.232 usage;
- open architecture where extensions/Packages definition work on the interface may be carried out;

¹ See *3GPP TS 23.002 V9.2.0 (2009-12)*.

- dynamic sharing of IMS-MGW physical node resources (a physical IMS-MGW can be partitioned into logically separate virtual MGWs/domains consisting of a set of statically allocated terminations);
- dynamic sharing of transmission resources between the domains as the IMS-MGW controls bearers and manage resources according to the H.248 protocols and functions for IMS.

The *Mn* reference point is internal to the *Optranex 248* switch and is, therefore, not exposed.

Mg Interface (MGCF to CSCF)

The *Mg* reference point allows the MGCF to forward incoming session signaling (from the circuit switched network) to the CSCF for the purpose of interworking with circuit switched networks. The protocol used for the *Mg* reference point is SIP (as defined by *RFC 3261*, other relevant RFCs, and additional enhancements introduced to support 3GPP's needs).

Mr Interface (S-CSCF to MRFC)

The *Mr* reference point allows interaction between an S-CSCF and an MRFC. The protocol used for the *Mr* reference point is SIP (as defined in *RFC 3261*, other relevant RFCs, and additional enhancements introduced to support 3GPP's needs).

Mp Interface (MRFC to MRFP)

The *Mp* reference point allows an MRFC to control media stream resources provided by an MRF. The *Mp* reference point has the following properties:

- full compliance with the H.248 standard; and,
- open architecture where extensions (packages) definition work on the interface may be carried out.

The *Mp* reference point is internal to the *Optranex 248* switch and is, therefore, not exposed.

Mj Interface (BGCF to MGCF)

The *Mj* reference point allows the Breakout Gateway Control Function to exchange session signaling messages with the Media Gateway Control Function for the purpose of interworking to the circuit-switched networks, or for transit networks. The *Mj* reference point is based on external specifications (i.e. SIP).

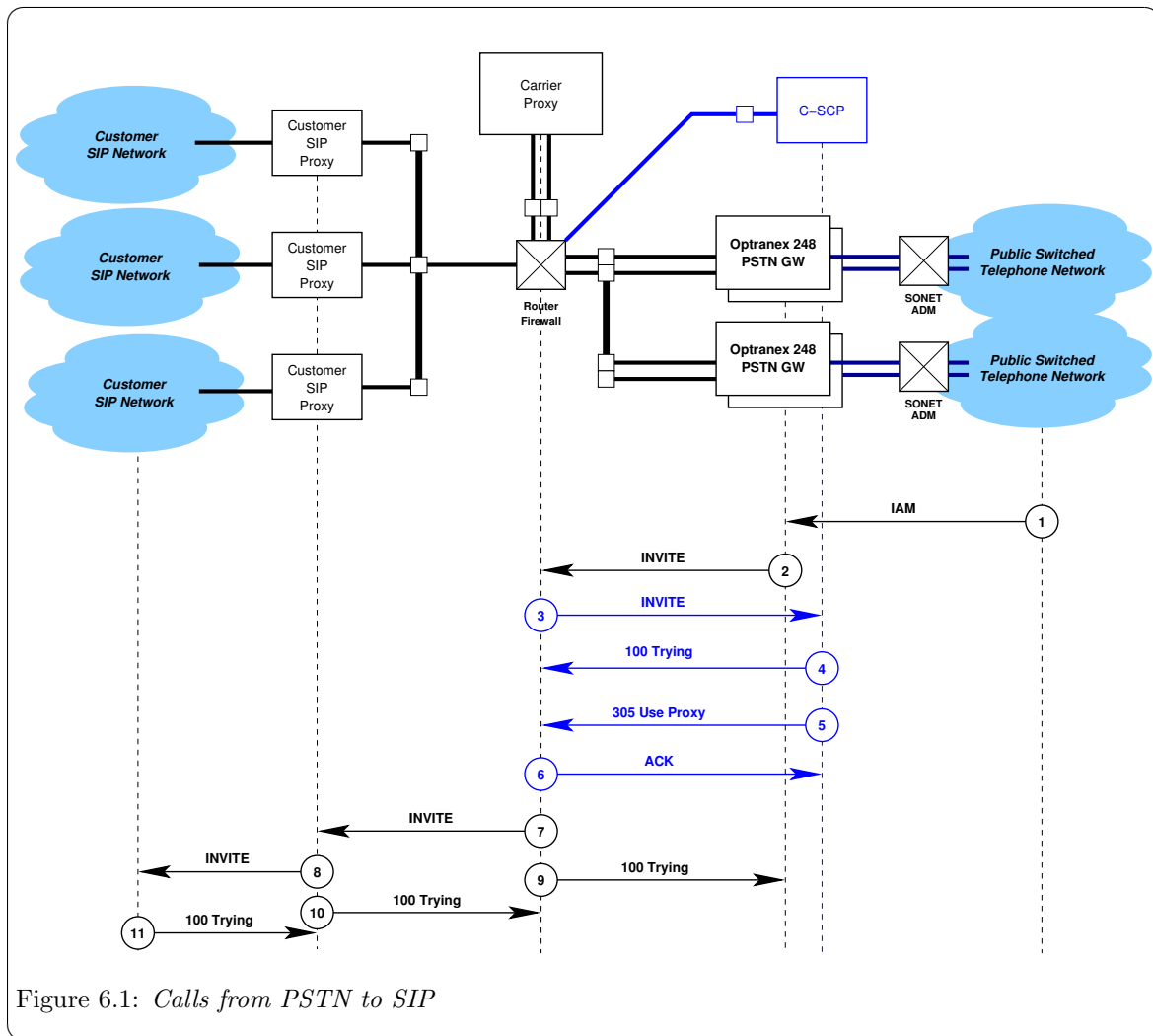
6 Protocol Architecture

6.1 C-SCP Interworking Call Flows

The following sections provide interworking call flows between the *Optranex 248* switch and the *C-SCP* platform.

6.2 Calls from PSTN to SIP

Calls from the PSTN terminating on the SIP network that are processed by the *Optranex 248* switch and further processed by the *C-SCP* platform are illustrated in [Figure 6.1](#).



1. Point (1) in [Figure 6.1](#).

A call is originated and terminates from the PSTN on the *Optranex 248* switch. An ISUP IAM is received by the *Optranex 248* switch. For an example of the IAM, see [Example 6.1](#).

<i>FCI</i>	number translated
<i>CgPN</i>	NSN, 222-333-4444, presentation-restricted
<i>CdPN</i>	NSN, 777-666-5555
<i>JIP</i>	830-660
<i>GAP</i>	ported number, NSN, 999-666-5555
<i>GAP</i>	dialed number, NSN, 444-444-4444, presentation-allowed
<i>GAP</i>	additional user provided number - not verified, NSN, 666-666-6666, presentation-allowed
<i>GN</i>	calling name, name available/unknown, "Joe", presentation-allowed
<i>GN</i>	original called name, name available/unknown, "Sue", presentation-allowed
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0

Example 6.1: *IAM received by Optranex 248 in Successful PSTN to SIP Call*

2. Point (2) in [Figure 6.1](#).

The information from the received IAM is translated to a SIP INVITE message and the INVITE is sent to the *Carrier Proxy*.

Calls coming from the PSTN will have information translated from the IAM to a SIP INVITE and the INVITE will be sent to the *Carrier Proxy* for number analysis and customer proxy selection or treatment.

All fields in the INVITE (Request-URI, To, From, P-Asserted-Identity, and P-Requested-Identity) will have a **tel** URI formatted number.

The INVITE corresponding to the above IAM is listed in [Message 6.1](#).

```
INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001
Max-Forwards: 33
From: "Joe" <tel:+16666666666>;tag=7643kals
To: "Sue" <tel:+14444444444>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:gw1@heramax.com:5060;transport=udp>
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
Content-Type: application/sdp
Content-Length: 134

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Message 6.1: *Optranex 248 INVITE generated from IAM in Successful PSTN to SIP Call*

For more detail concerning the mapping of ISUP parameters to the outgoing INVITE method, see [\[undefined\]](#) [\[undefined\]](#), page [\[undefined\]](#) and [\[undefined\]](#) [\[undefined\]](#), page [\[undefined\]](#).

3. Point (3) in [Figure 6.1](#).

The *Carrier Proxy* forwards the INVITE as a stateful proxy to the *C-SCP*.

```

INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
Max-Forwards: 32
From: "Joe" <tel:+16666666666>;tag=7643kals
To: "Sue" <tel:+14444444444>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:gw1@heramax.com:5060;transport=udp>
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
Content-Type: application/sdp
Content-Length: 134

```

```

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.2: *Carrier Proxy INVITE in Successful PSTN to SIP Call*

4. Point (4) in [Figure 6.1](#).

If the *C-SCP* platform expects that it will take longer than 200 milliseconds to respond to an INVITE with a 3xx, 4xx or 5xx response, it should send a 100 Trying message in response to the INVITE. Should the *C-SCP* platform delay in responding to the INVITE, the *Optranex 248* or *Carrier Proxy* might cancel the INVITE with a CANCEL request for the dialog (should the call be released on the ISUP side before the response is received). The *C-SCP* platform is expected to acknowledge the CANCEL request and not further respond to the corresponding INVITE.

```

SIP/2.0 100 Trying
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
CSeq: 1 INVITE
Content-Length: 0

```

Message 6.3: *100 Trying Response from C-SCP for Successful PSTN to SIP Call*

5. Point (5) in [Figure 6.1](#).

The *C-SCP* platform will analyze the information and either return a 305 Use Proxy message directing the *Carrier Proxy* to send the call to a customer proxy, or will return a 4xx or 5xx

message indicating an error with a **Reason** header indicating the *Q.850* or *ANSI* cause value according to the logic outlined later under unsuccessful call flows.

The *C-SCP* platform, for successful calls, sends a **305 Use Proxy** message in response to the *Carrier Proxy* with the customer proxy to which to forward the call in the **Contact** header. The procedure for analyzing the received **INVITE** is detailed in [\[undefined\]](#), page [\[undefined\]](#).

An example using DNS SRV entries is as follows:

```
SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=7643kals
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
CSeq: 1 INVITE
Contact: "Sue Zy" <sip:sue@customer1.com>
P-Asserted-Identity: Joe Bloe
<tel:+12223334444;cpc=ordinary;oli=00;rn+=1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
Content-Length: 0

Message 6.4: 305 Use Proxy from C-SCP for Successful PSTN to SIP Call
```

For other mechanisms for distributing calls over customer proxies, see [\[undefined\]](#), page [\[undefined\]](#).

6. Point (6) in [Figure 6.1](#).

The *Carrier Proxy* acknowledges the response with an **ACK** message.

```
ACK sip:tellingua@heramax.com SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA.1
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
Max-Forwards: 70
From: "Joe" <tel:+16666666666>;tag=7643kals
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
CSeq: 1 ACK
Content-Length: 0

Message 6.5: Optranex 248 ACK Request for Successful PSTN to SIP Call
```

7. Point (7) in [Figure 6.1](#).

Information from the **305 Use Proxy** response from the *C-SCP* platform (primarily the **Contact** headers) is used to formulate a new **INVITE** message that is then sent to the customer proxy. For more details, see [\[undefined\]](#), page [\[undefined\]](#).

8. Point (8) in [Figure 6.1](#).

The customer proxy then forwards the **INVITE** message within the customer's SIP network.

Additional actions beyond point (8) in [Figure 6.1](#) are in accordance with *3GPP TS 29.163 V9.0.0 (2009-12)* and the interactions between the *MGCF* and *I-CSCF* or *S-CSCF* as described in *3GPP TS 24.229 V9.2.0 (2009-12)* and do not further involve the *C-SCP* platform.

6.2.1 Translation of IAM to INVITE

The translation from the IAM to the INVITE at the *Optranex 248* switch, sent to the *Carrier Proxy* will be in accordance with *3GPP TS 29.163 V9.0.0 (2009-12) 7.2.3.2.2.*, summarized and with additional features as follows:

6.2.1.1 B Number

The **Request-URI** and **To** field will always contain the non-porting or porting called number.

```
INVITE tel:+19996665555 SIP/2.0
To: <tel:+19996665555>
...
```

If the *IAM Forward Call Indicators* indicates **number translated** the *npdi* parameter will be added to the **Request-URI** (but not the **To** header).

```
INVITE tel:+19996665555;npdi SIP/2.0
To: <tel:+19996665555>
...
```

If the *IAM Generic Address Parameter* parameter contains a porting number it will be the number in the **To** header and the **Request-URI**. The *Called Party Number* (which in this case is the LRN) will be added to the **Request-URI** (but not the **To** header) in the *rn* parameter field.¹

```
INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
To: <tel:+19996665555>
...
```

If the *IAM Generic Name* parameter is present and contains a **original called name** that is **presentation-allowed**, it will be used in the **display-name** portion of the **To** field. If the name is not available, the **display-name** "Unavailable" will be used instead.²

```
INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
To: "Joe" <tel:+19996665555>
...
```

6.2.1.2 A Number

If the *Calling Party Number* is not present, or the *Calling Party Number* is **presentation-restricted**, and there is no *Generic Address Parameter additional user provided number*, the **From** header will contain:

```
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=XXX
```

If the *Calling Party Number* is present, but is **presentation-restricted**, the "id" token will be added to a **Privacy** header.

```
Privacy: id
```

If the *Calling Party Number* is present and is **presentation-allowed**, or there is a *Generic Address Parameter additional user provided number* that is **presentation-allowed**, the **From** header will contain the *tel URL* formatted number:

```
From: <tel:+14443332222>;tag=XXX
```

¹ See *3GPP TS 29.163 7.2.3.2.2A*.

² See *3GPP TS 29.163 7.2.3.2.2.3*.

If there is a *Calling Party Number* present (whether **presentation-allowed** or **presentation-restricted**, as long as it is network-provided and verified, it will be placed in the **P-Asserted-Identity** header. This behaviour is consistent with that described for the *MGCF* in *3GPP TS 29.163 7.2.3.2.2*.

P-Asserted-Identity: <tel:+14443332222>

Otherwise, if no network-provided or verified *Calling Party Number* is present, the **P-Asserted-Identity** header will contain the invalid number (000-000-0000).³

P-Asserted-Identity: <tel:+10000000000>

If a *Jurisdiction Information Parameter* parameter is present in the **IAM**, it will be added in the *rn* field of the **P-Asserted-Identity** header (but not the **From** header). This behaviour is consistent with *RFC 3325* and *RFC 4694*.⁴

P-Asserted-Identity: <tel:+10000000000;rn=+1860660>

The incoming trunk group will always be placed in the **P-Asserted-Identity** header (but not the **From** header) and can be used to determine the *Jurisdiction Information Parameter* if one is present. This behaviour is consistent with *RFC 3325* and *RFC 4904*.

P-Asserted-Identity: <tel:+10000000000;rn=+1860660;trgp=7654321>

If there is a *Calling Party Category* parameter in the **IAM**, it will be added to the **P-Asserted-Identity** (but not the **From** header) in the *cpc* parameter. This behaviour is consistent with that described for the *MGCF* in *3GPP TS 29.163 Annex C.2*.⁵

P-Asserted-Identity:
<tel:+14443332222;cpc=ordinary;oli=00;rn=+1860660
;trgp=EDTNAB01DS1;trunk-context=gw1.heramax.com>

If there is an *Originating Line Information* parameter in the **IAM**, it will be added to the **P-Asserted-Identity** (but not the **From** header) in the *oli* parameter. This behaviour is consistent with that described for the *MGCF* in *3GPP TS 29.163 Annex H.2*.⁶

P-Asserted-Identity:
<tel:+14443332222;cpc=ordinary;oli=00;rn=+1860660
;trgp=EDTNAB01DS1;trunk-context=gw1.heramax.com>

If there is a *Generic Name* parameter in the **IAM** containing the **calling name** (ISUP method) and it is **presentation-allowed**, it will be used as the **display-name** of both the **From** and **P-Asserted-Identity** URIs. When it is **presentation-restricted** it will only be used for the **P-Asserted-Identity** URI and privacy will be elevated to ‘user’. If the name is not available, the **display-name** “Unavailable” will be used. This behavior is consistent with that described for the *MGCF* in *3GPP TS 29.163*.⁷

³ Inserting an invalid number in the *user-info* portion of the **tel** URI is in variation from *3GPP TS 29.163*, which has no method for passing calling party parameters without a calling party number.

⁴ See *RFC 4694* Section 5.2.4. Note, however, that neither ITU-T Q.1912.5/1998 nor *3GPP TS 29.163 V9.0.0* support the concept of communicating the *Jurisdiction Information Parameter* from the **IAM** to the **INVITE**.

⁵ See *3GPP TS 29.163 V9.0.0 (2009-12) Annex C.2*, *RFC 3325* and *draft-patel-dispatch-cpc-oli-parameter-02*. See also, *3GPP TS 29.163 7.2.3.2.2.3A*.

⁶ See *3GPP TS 29.163 V9.0.0 (2009-12) Annex H.2*, *RFC 3325* and *draft-patel-dispatch-cpc-oli-parameter-02*. See also, *3GPP TS 29.163 7.2.3.2.2.3B*.

⁷ See *3GPP TS 29.163 V9.0.0*, *ITU-T Rec. Q.1912.5* and *RFC 3325*.

```

From: Unavailable <sip:anonymous@anonymous.invalid>;tag=XXX
P-Asserted-Identity: Unavailable
  <tel:+14443332222;cpc=ordinary;oli=00;rn=+1860660
    ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id

```

```

From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=XXX
P-Asserted-Identity: Joe
  <tel:+14443332222;cpc=ordinary;oli=00;rn=+1860660
    ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: user

```

6.2.1.3 Charging Information

Charging information from the received IAM will be translated to a **P-Charge-Info** header added to the resulting INVITE in accordance with *draft-york-sipping-p-charge-info-07* with the following considerations:

- *draft-york-sipping-p-charge-info-07* shows only **sip** URIs (and crude ones without a *user=phone* parameter); however, the ABNF syntax of the **P-Charge-Info** header allows an *addr-spec* to be included in the header. The *Optranex 248* platform will always place a **tel** URI in the **P-Charge-Info** header.

Whenever a **P-Charge-Info** header is included in the resulting INVITE, the incoming trunk group will always be placed in the **P-Charge-Info** header (in addition to the **P-Asserted-Identity** header in the *tgrp* parameter of the **tel** URI. This placement of the *tgrp* parameter in a **tel** URI is in accordance with *RFC 4904*. This incoming trunk group can be used to determine a *Jurisdiction Information Parameter* when one was not present in the received IAM and none was assigned to the incoming trunk group at the *Optranex 248*.

If there is a *Charge Number* parameter present in the received IAM, a **P-Charge-Info** header will be added to the resulting INVITE in accordance with *draft-york-sipping-p-charge-info-07*. The address digits (when present) of the *Charge Number* will be provided in the *telephone-subscriber* portion of a **tel** URI in the **P-Charge-Info** header. When address digits are not present in the *Charge Number* parameter (because the nature-of-address value indicates that no address is provided) either the *Calling Party Number* or *Called Party Number* parameter will be used as the source of the address digits. An *npi* parameter will be added to the header with the value **ISDN** in accordance with *draft-york-sipping-p-charge-info-07*. A *noa* parameter will also be added to the header with the nature-of-address value contained in the *Charge Number* parameter, also in accordance with *draft-york-sipping-p-charge-info-07*. Because a *Originating Line Information* parameter must also be present whenever a *Charge Number* is present in a received ANSI IAM, the *Originating Line Information* parameter will be added to the **tel** URI in the **P-Charge-Info** header (in addition to the **P-Asserted-Identity** header). This use of the *oli* parameter in a **tel** URI is in accordance with *3GPP TS 29.163* and *draft-patel-dispatch-cpc-oli-parameter-02*.

```

P-Charge-Info: <tel:+13333333333;oli=00
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN;noa=3

```

If there is an *Originating Line Information* parameter present in the received IAM, but no *Charge Number*, the *Calling Party Number* will instead be used to populate the *telephone-subscriber* portion of the **tel** URI in the **P-Charge-Info** header in accordance with *ANSI T1.113/2000*.⁸ A *npi* parameter will also be added to the **P-Charge-Info** header with the value **ISDN**. No *noa* parameter

⁸ Note that the *Called Party Number* is always present and the *Calling Party Number* must be present in an ANSI IAM when the *Originating Line Information* parameter is present and no *Charge Number* is present.

will be added to the **P-Charge-Info** header. This addition of the *npi* and *noa* parameters is in accordance with *draft-york-sipping-p-charge-info-07*. For example:

```
P-Charge-Info: <tel:+12223334444;oli=00
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
```

If a *Jurisdiction Information Parameter* is also present in the received **IAM**, or a *Jurisdiction Information Parameter* value is associated with the incoming trunk group, it will be added as a *rn* parameter in the **tel** URI of any **P-Charge-Info** header (in addition to the *P-Asserted-Identity* header). This use of the *rn* parameter to represent *JIP* in **tel** URIs associated with the origination is in accordance with *RFC 4694*.

```
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
```

6.2.1.4 Example IAM to INVITE Translation

For this example the **IAM** has the contents shown in [Example 6.2](#).

<i>FCI</i>	number translated
<i>CgPN</i>	NSN, 222-333-4444, presentation-restricted
<i>CdPN</i>	NSN, 777-666-5555
<i>JIP</i>	830-660
<i>GAP</i>	ported number, NSN, 999-666-5555
<i>GAP</i>	dialed number, NSN, 444-444-4444, presentation-allowed
<i>GAP</i>	additional user provided number - not verified, NSN, 666-666-6666, presentation-allowed
<i>GN</i>	calling name, name available/unknown, "Joe", presentation-allowed
<i>GN</i>	original called name, name available/unknown, "Sue", presentation-allowed
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0

Example 6.2: *IAM for IAM to INVITE Translation Example*

The **INVITE** corresponding to the **IAM** in [Example 6.2](#) is shown in [Message 6.6](#).


```

INVITE tel:+999666555;npdi;rn=+1777666555 SIP/2.0
Via: SIP/2.0/UDP sip:gw1.heramax.com:5060;branch=z9hG4bK-000001
Max-Forwards: 33
From: Joe <tel:+16666666666>;tag=7643kals
To: Sue <tel:+4444444444>
CSeq: 1 INVITE
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
Contact: <sip:+16666666666
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com
@gw1.heramax.com;user=phone>
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Type: application/sdp
Content-Length: 134

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IPV4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.6: *Optranex 248 INVITE from IAM for Successful PSTN to SIP Call Example*

Notes for [Message 6.6](#):

1. Request-URI

The *telephone-subscriber* portion of the Request-URI contains the ported number from the GAP parameter. The *npdi* parameter is present in the Request-URI indicating that the number was dipped by the originator, and the *rn* parameter is present in the Request-URI and contains the local routing number.

2. Max-Forwards

The value of the Max-Forwards header is calculated by applying a weighting factor to the ISUP *Hop Counter* parameter.

3. From

The *display-name* portion of the From header contains the **presentation-allowed** calling name from the *Generic Name* ISUP parameter. The *telephone-subscriber* portion of the From header contains the **presentation-allowed** additional user provided number from the *Generic Address Parameter* ISUP parameter.

4. To

The *display-name* portion of the To header contains the **presentation-allowed** original called name from the *Generic Name* ISUP parameter. The *telephone-subscriber* portion of the To header contains the **presentation-allowed** dialled number from the *Generic Address Parameter* ISUP parameter.

5. P-Asserted-Identity

The *display-name* portion of the P-Asserted-Identity header contains the **presentation-allowed** calling name from the *Generic Name* ISUP parameter. The *telephone-subscriber*

portion of the **P-Asserted-Identity** header contains the **presentation-restricted** calling party number from the *Calling Party Number* ISUP parameter.

6. Privacy

The **Privacy** header contains the *id* token because the *telephone-subscriber* portion of the **P-Asserted-Identity** header is **presentation-restricted**. The privacy level is set to *id* instead of *user* because the *display-name* portion of the **P-Asserted-Identity** header is **presentation-allowed**.

7. P-Charge-Info

The **P-Charge-Info** header is added to the message because an *Originating Line Information* parameter is present in the received **IAM**. The *telephone-subscriber* portion of the **tel** URI in the header is taken from the *Calling Party Number* because a *Charge Number* does not appear in the **IAM**. The *oli* and *tgrp* parameter in the **tel** URI in the **P-Charge-Info** is always populated by the *Optranex 248*. Because a *Jurisdiction Information Parameter* is present in the received **IAM**, a *rn* parameter is populated in the **tel** URI. The *npi* header parameter has the value **ISDN** because the numbering plan of the *Calling Party Number* is E.164.

8. SDP

The SDP specification for the call is calculated by mapping the incoming ISUP circuit from the CS domain to a fixed IP address (host name) and RTP port number on the IMS CN side. The remainder of the specification is for PCM mu-law 8000 samples per second audio (speech).

6.2.2 Processing of Calls Terminating from the PSTN

The requirements for *C-SCP* processing of the **INVITE** from the *Carrier Proxy* for calls terminating from the PSTN are described in this subsection.

6.2.2.1 Extracting Information from the INVITE

The *C-SCP* platform should largely ignore the **From** and **To** headers in the **INVITE** (except, of course, for the *tag* parameter).

For information concerning the termination of the call (B Number), the *C-SCP* platform should examine the *Request-URI* instead of the **To** header of the **INVITE**.

For information concerning the originator (A Number) of the call, the *C-SCP* platform should examine the **P-Asserted-Identity** header instead of the *From* header of the **INVITE**.

For information concerning the charging (Charge Information) of the call, the *C-SCP* platform should examine the **P-Charge-Info** header instead of the *From* header of the **INVITE**.

The *C-SCP* platform should determine the called party parameters (B number) from the **Request-URI** of the **INVITE**. The minimum information that will be present in the **Request-URI** will be the *Called Party Number*. The most information that will be in the **Request-URI** will be the *Called Party Number*, **number translated** (NP dip) indication and *Local Routing Number*. The called party name may also be present in the *display-name* field of the **To** header. See [Table 6.1](#) for details.

ISUP Parameter	Request-URI Field	Reference
Called Party Number	telephone-subscriber	RFC 3966
Number Translation	npdi parameter	RFC 4694
Local Routing Number	rn= parameter	RFC 4694
Called Party Name	display-name	RFC 3966

Table 6.1: *Request-URI Parameter Mapping*

The *C-SCP* platform should determine the calling party parameters (A number) from the **P-Asserted-Identity** header of the INVITE. The minimum information present will be the incoming trunk group number. The most information present will be the calling name, calling number, calling party's category, originating line information, jurisdiction information parameter and incoming trunk group number. See [Table 6.2](#) for details.

ISUP Parameter	P-Asserted-Identity Field	Reference
Incoming Trunk Group	tgrp= parameter	RFC 4904
Calling Name	display-name	RFC 3966
Calling Party Number	telephone-subscriber	RFC 3966
Calling Party's Category	cpc= parameter	TS 29.163 Annex C
Originating Line Information	oli= parameter	TS 29.163 Annex H
Jurisdiction Information Parameter	rn= parameter	RFC 4694

Table 6.2: *P-Asserted-Identity Parameter Mapping*

The *C-SCP* platform should determine the charging information from the **P-Charge-Info** header of the INVITE. The minimum information present (when the header is present) will be the charge number, originating line information, and incoming trunk group number. The most information present will be the charge number, originating line information, incoming trunk group number, numbering plan indicator and nature of address indicator value. See [Table 6.3](#) for details.

ISUP Parameter	P-Charge-Info Field	Reference
Incoming Trunk Group	tgrp= parameter	RFC 4904
Charge Number Calling Party Number Called Party Number	telephone- subscriber	RFC 3966
Originating Line Information	oli= parameter	TS 29.163 Annex H
Jurisdiction Information Parameter	rn= parameter	RFC 4694
–	npi=ISDN parameter	draft-york-sipping-p-charge-info-07
Charge Number	noa= parameter	draft-york-sipping-p-charge-info-07

Table 6.3: *P-Charge-Info Parameter Mapping*

6.2.2.2 305 Use Proxy Response

Using the B number parameters in the **Request-URI** of the **INVITE** as listed in [Table 6.1](#), (and possibly using the A number parameters in the **P-Asserted-Identity** header as listed in [Table 6.2](#), and perhaps the charge information parameter in the **P-Charge-Info** header as listed in [Table 6.3](#)), the *C-SCP* platform should determine the customer proxy (or proxies) to which to route the call and return them in a **305 Use Proxy** or **303 Proxy Redirect** response to the **INVITE**.

The **Contact** header(s) contained in the **305 Use Proxy** or **303 Proxy Redirect** response must be formatted as **sip** URIs.

A possible response to the example above is shown in [Message 6.7](#).

```
SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+19996665555@sansay1.customer1.com;user=phone>
Contact: <sip:+19996665555@sansay2.customer2.com;user=phone>
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
; tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
; tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Length: 0
Message 6.7: 305 Use Proxy Response for Successful PSTN to SIP Call – Explicit Proxy
```

Depending on preferred network configuration, the *C-SCP* can either directly address customer proxies in the **Contact** headers of the **305 Use Proxy** or **303 Proxy Redirect** response (as shown in [Message 6.7](#)), or can use sip DNS SRV records for the domain by specifying the general domain, as shown in [Message 6.8](#).

```

SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+19996665555@customer1.com;user=phone>
P-Asserted-Identity: Joe
  <tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Length: 0
Message 6.8: 305 Use Proxy Response for Successful PSTN to SIP Call – DNS SRV Approach

```

The *C-SCP* may also do an ENUM query to determine the user within the customer domain similar to that shown in [Message 6.9](#).

Note that in this approach, the **Contact** header no longer contains a **tel** URI formatted *user-info* portion in the **sip** URI, but contains an actual **sip** user name. Note that this *will* affect the **Request-URI** in the INVITE message ultimately launched to the Customer Proxy, but, in accordance with *RFC 3261*, the **To** field will remain unaltered.

```

SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:sue@customer1.com>
P-Asserted-Identity: Joe
  <tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Length: 0
Message 6.9: 305 Use Proxy Response for Successful PSTN to SIP Call – I-ENUM Query Approach

```

As an option, the *C-SCP* may also query a CNAM database and determine the calling name for the *Calling Party Number* and alter the **P-Asserted-Identity** to reflect the calling party name, as shown in [Message 6.10](#).

```

SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:sue@customer1.com>
P-Asserted-Identity: Joe Bloe
<tel:+12223334444;cpc=ordinary;oli=00;rn+=1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn+=1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Length: 0
Message 6.10: 305 Use Proxy Response for Successful PSTN to SIP Call – CNAM Query Approach

```

6.2.2.3 Processing of the 305 Use Proxy Response

The *Carrier Proxy* switch may adjust the **To** and **From** fields on the INVITE sent to the customer proxy. The **To** header may be adjusted from information in the **Contact** header of the 305 Use Proxy response; the **From** header may be adjusted from information in the **P-Asserted-Identity** header of the 305 Use Proxy response.

The values in the **Contact** header or headers constitute the entire target set for the gateway. Acting as a proxy, the *Carrier Proxy* selects members in the target set and for each performs the following (in accordance with *draft-ietf-speermint-srv-naptr-use-06*, *RFC 2915* and *RFC 2782*):

1. A DNS NAPTR query is performed on the domain name portion of the **sip** URI. If there are records in the result, the ‘_sip._udp.*’ record is selected. Otherwise, if there are no records in the result, a new name is generated by concatenating ‘_sip._udp.’ and the domain name.
2. A DNS SRV query is performed on the name resulting from the previous step. If there are records in the result, a record is selected according to the rules of DNS SRV entries. Otherwise, the original domain name portion of the **sip** URI selected as the new name.
3. If the result from the previous step is an A or AAAA RR, this step is skipped and the IP address is used to address the INVITE to the customer proxy. If the result from the previous step is a name, a DNS QUERY query is performed on the name.

6.2.2.4 Example of a Successful Call from PSTN to SIP

Refer to [Figure 6.1](#) for the messages exchanged that correspond with the numbers below.

1. Point (1) in [Figure 6.1](#).

A call is generated within the PSTN and terminates on the *Optranex 248* switch. The *Optranex 248* switch receives an IAM message, which, in this example, contains the parameters shown in [Example 6.3](#).

<i>FCI</i>	number translated
<i>CgPN</i>	NSN, 222-333-4444, presentation-restricted
<i>CdPN</i>	NSN, 777-666-5555
<i>JIP</i>	830-660
<i>GAP</i>	ported number, NSN, 999-666-5555
<i>GAP</i>	dialed number, NSN, 444-444-4444, presentation-allowed
<i>GAP</i>	additional user provided number - not verified, NSN, 666-666-6666, presentation-allowed
<i>GN</i>	calling name, name available/unknown, "Joe", presentation-allowed
<i>GN</i>	original called name, name available/unknown, "Sue", presentation-allowed
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0

Example 6.3: *IAM received for Successful PSTN to SIP Call Example*

2. Point (2) in [Figure 6.1](#).

The *Optranex 248* switch uses the information from the received *IAM* to generate an *INVITE* message which is sent to the *Carrier Proxy*. The *INVITE* corresponding to the *IAM* shown in [Example 6.3](#) is shown in [Message 6.11](#).

```

INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+16666666666
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com
@gw1.heramax.com:5060;user=phone>
Max-Forwards: 33
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Type: application/sdp
Content-Length: 134

```

```

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.11: *INVITE generated for Successful PSTN to SIP Call Example*

3. Point (3) in [Figure 6.1](#).

The *Carrier Proxy* forwards the *INVITE* message to the *C-SCP* for processing.

```

INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
Max-Forwards: 32
From: "Joe" <tel:+16666666666>;tag=7643kals
To: "Sue" <tel:+14444444444>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:gw1@heramax.com:5060;transport=udp>
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
Content-Type: application/sdp
Content-Length: 134

```

```

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.12: *INVITE forwarded in Successful PSTN to SIP Call Example*

4. Point (4) in [Figure 6.1](#).

In this example, the *C-SCP* platform determines that it might not respond to the *INVITE* message within 200 milliseconds and so sends a 100 *Trying* message as shown in [Message 6.13](#).

```

SIP/2.0 100 Trying
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Content-Length: 0

```

Message 6.13: *100 Trying Response in Successful PSTN to SIP Call Example*

5. Point (5) in [Figure 6.1](#).

The *C-SCP* platform must perform the processing described in this document (see [\(undefined\)](#) [(undefined)], page (undefined)) and, when the call is successful, return a 305 *Use Proxy* message containing the customer proxy or proxies to which to redirect the call.


```

SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:sue@customer1.com>
P-Asserted-Identity: Joe Bloe
  <tel:+12223334444;cpc=ordinary;oli=00;rn+=1830660
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn+=1830660
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>;npi=ISDN
Content-Length: 0

Message 6.14: 305 Use Proxy Response in Successful PSTN to SIP Call Example

```

Note that in [Message 6.14](#), the **Contact** header is translated with an I-ENUM query to the actual sip address of the termination and uses the generic domain usable in the NAPTR and SRV entry approach.

6. Point (6) in [Figure 6.1](#).

The *Carrier Proxy* switch acknowledges the 305 Use Proxy response with an ACK request as shown in [Message 6.15](#).

```

ACK sip:anonymous@gw1.heramax.com SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 ACK
Contact: <sip:+16666666666
  ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com
  @gw1.heramax.com:5060;user=phone;transport=udp>
Content-Length: 0

Message 6.15: ACK Request in Successful PSTN to SIP Call Example

```

7. Point (7) in [Figure 6.1](#).

The *Carrier Proxy* switch generates an INVITE message toward the specified customer proxy. The customer proxy is not within the *MGCF* trust domain and, therefore, all privacy fields are stripped. The INVITE message is shown in [Message 6.16](#).

```

INVITE sip:sue@customer1.com SIP/2.0
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001
From: "Joe" <tel:+16666666666>;tag=XXX
To: "Sue" <tel:+14444444444>
Call-ID: 111223@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+16666666666
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com
@gw1.heramax.com:5060;user=phone;transport=udp>
Max-Forwards: 33
Content-Type: application/sdp
Content-Length: 134

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.16: *Customer Proxy INVITE for Successful PSTN to SIP Call Example*

8. Point (8) in [Figure 6.1](#) and beyond.

The *C-SCP* platform is no longer involved in the call. Interaction between the *Carrier Proxy* and the customer proxy is in accordance with the exchange between the *MGCF* and *I-CSCF* or *S-CSCF* in *3GPP TS 24.229* and *3GPP TS 29.163* (and RFCs called out by these specification) with minor variations for US networks.

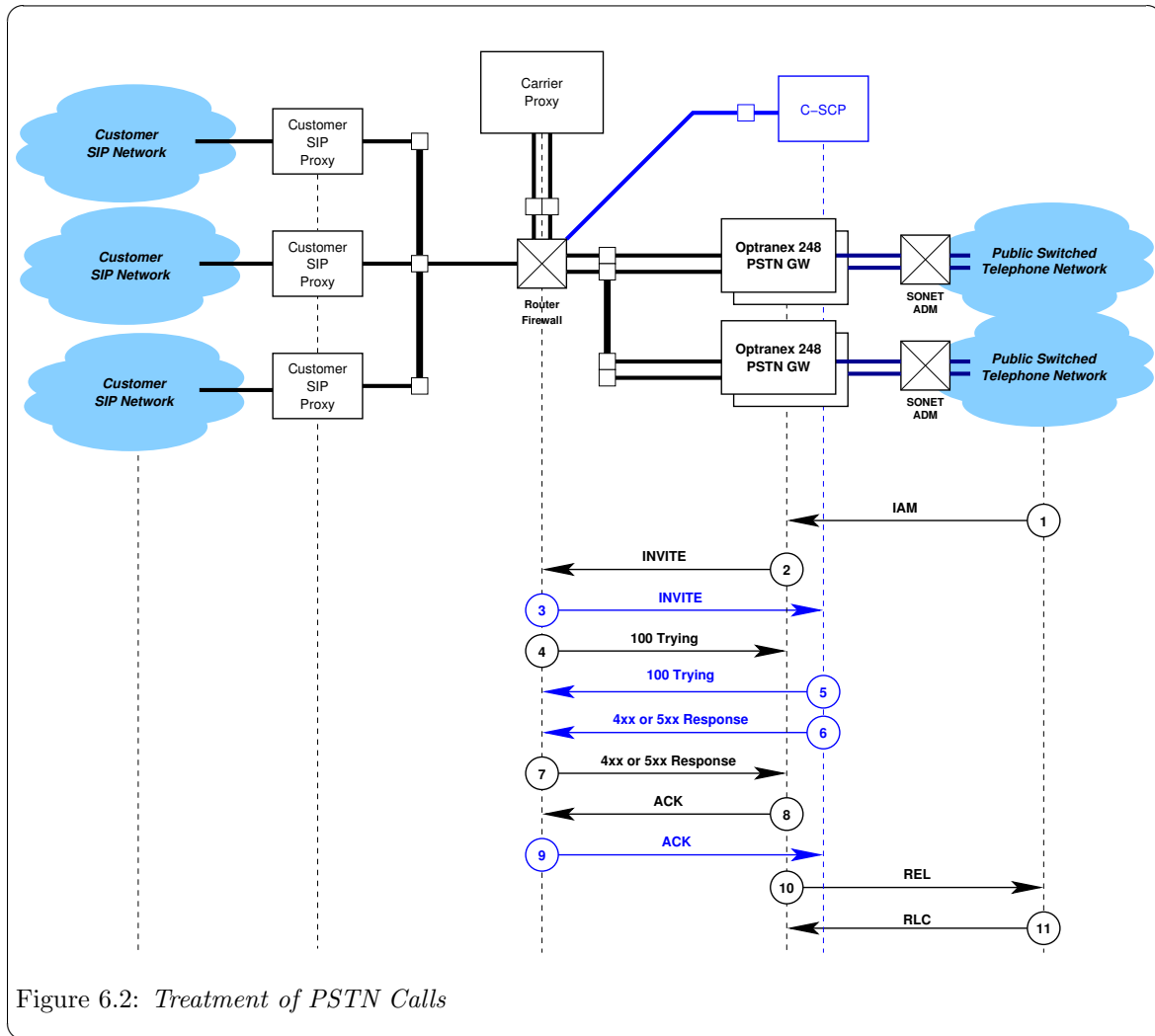
6.2.3 Treatment of PSTN Originated Calls at the C-SCP Platform

Because the *Optranex 248* switch is not populated with subscriber data⁹ and, therefore, cannot perform number analysis, and also because the *Optranex 248* switch does not perform LNP dips,¹⁰ the *C-SCP* must perform part of the normal ISUP processing and distinguish to the *Optranex 248* switch various digit analysis failures so that the *Optranex 248* switch can apply the necessary treatment of the call. This can in some cases be performed with a selection of a response code to the INVITE sent to the *C-SCP* platform, in other cases must be distinguished using the **Reason** header.¹¹

⁹ The *Optranex 248* switch is not being populated with customer data for the trial: it is capable of populating customer data and forming dial plans from such data, particularly using the ENUM approach illustrated in [Section 6.4 \[ENUM Calls from PSTN to SIP\]](#), page 83.

¹⁰ The *Optranex 248* switch will not perform LNP dips for the trial: the switch is capable of performing TCAP LNP dips or I-ENUM number portability dips in normal operation as illustrated in [Section 6.5 \[ENUM Calls from SIP to PSTN\]](#), page 91.

¹¹ See RFC 3326 and draft-jesske-dispatch-reason-in-responses-01 for more information on the **Reason** header. Note that this header is also proscribed by *ITU-T Recommendation Q.1912.5* as well as *3GPP TS 29.163*.



The basic call flow for a unsuccessful (treated) call terminating from the PSTN is illustrated in Figure 6.2. The key features of the call flow are as follows:

1. Point (1) in Figure 6.2.

The PSTN has generated a call that terminates on the *Optranex 248*, and an IAM is launched from the PSTN to the *Optranex 248*. An example of the IAM is shown in Example 6.4.

2. Point (2) in Figure 6.2.

The *Optranex 248* translates the information from the received IAM to a SIP INVITE and sends it to the *Carrier Proxy* platform for processing. The translation of the information from the received IAM into the sent SIP INVITE was detailed previously in [undefined], page undefined. An example of the INVITE is shown in Message 6.28.

3. Point (3) in Figure 6.2.

The *Carrier Proxy* forwards the INVITE to the *C-SCP* for further processing. An example of the INVITE message is shown in Message 6.29.

4. Point (4) in Figure 6.2.

As the *Carrier Proxy* has forwarded the INVITE message and is no longer in control of when a response will be made, the *Carrier Proxy* responds with a 100 Trying message to the *Optranex 248*. An example of the INVITE message is shown in [Message 6.30](#).

Should the *Carrier Proxy* delay in responding to the INVITE, the *Optranex 248* might cancel the INVITE with a CANCEL request for the dialog (should the call be released before a response is forthcoming). In the event of a CANCEL request and not further respond to the corresponding INVITE.

5. Point (5) in [Figure 6.2](#).

If the *C-SCP* platform expects that it will take longer than 200 milliseconds to respond to an INVITE with a 3xx, 4xx or 5xx response, it should send a 100 Trying message in response to the INVITE.

Should the *C-SCP* platform delay in responding to the INVITE, the *Optranex 248* might cancel the INVITE with a CANCEL request for the dialog (should the call be released before a response is forthcoming). In the event of a CANCEL, the *Carrier Proxy* and *C-SCP* platform are expected to acknowledge the CANCEL request and not further respond to the corresponding INVITE. An example of the 100 Trying message is shown in [Message 6.13](#).

6. Point (6) in [Figure 6.2](#).

The *C-SCP* platform analyzes the information in the received INVITE message and determines that treatment is necessary and sends a 4xx or 5xx response. The specific logic that the *C-SCP* platform uses to determine this, and the specific response chosen, is detailed later in [\[undefined\]](#) [\[undefined\]](#), page [\[undefined\]](#). An example of a 404 message is shown in [Message 6.31](#).

7. Point (7) in [Figure 6.2](#).

The *Carrier Proxy* returns the 4xx or 5xx response to the *Optranex 248* gateway switch. An example of a 404 message is shown in [Message 6.32](#).

8. Point (8) in [Figure 6.2](#).

The *Optranex 248* switch, upon receiving the 4xx or 5xx response, acknowledges that response by sending a ACK message to the *Carrier Proxy* in a acknowledgement. An example of the ACK message is shown in [Message 6.33](#).

9. Point (9) in [Figure 6.2](#).

The *Carrier Proxy*, upon receiving the ACK from the *Optranex 248* gateway switch, propagates the ACK message to the *C-SCP*. An example of the ACK message is shown in [Message 6.34](#).

10. Point (10) in [Figure 6.2](#).

The *Optranex 248* switch releases both the RTP and channel resources associated with the incoming call and issues an ISUP REL message to the previous PSTN switch with the appropriate cause value (from the Reason header of the received 4xx or 5xx response).

11. Point (11) in [Figure 6.2](#).

The previous PSTN switch acknowledges the release with an ISUP RLC message.

6.2.3.1 Treatment of PSTN Calls

At point (6) in [Figure 6.2](#), the *C-SCP* platform should examine the B Number information present in the Request-URI per [Table 6.1](#) (and possibly A number information present in the P-Asserted-Identity per [Table 6.2](#)) of the INVITE and take exception in accordance with *3GPP TS 29.163*, *ITU-T Q.1912.5* and *RFC 3326* as detailed below.

When the destination number (*telephone-subscriber* portion of the Request-URI) exists and is allocated to a customer, the *C-SCP* formulates a 305 Use Proxy response and places the customer

proxy (or proxies) in the **Contact** header of the 305 Use Proxy response as discussed earlier. An example is repeated in [Message 6.17](#).

```
SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:sue@customer1.com>
P-Asserted-Identity: Joe Bloe
<tel:+12223334444;cpc=ordinary;oli=00;rn+=1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn+=1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Content-Length: 0
```

Message 6.17: *305 Use Proxy Response for Successful PSTN to SIP Call Example*

Unallocated (unassigned) number

When a dipped non-ported number is not found (that is, there is a *npdi* parameter in the **Request-URI** but no *rn* parameter, and the *telephone-subscriber* portion of the **Request-URI** is not allocated to a customer) the *C-SCP* platform should respond with a 404 Not Found response with **Reason** headers as shown in [Message 6.18](#).

```
SIP/2.0 404 Not Found
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=1 ;text="unallocated (unassigned) number"
Content-Length: 0
```

Message 6.18: *Unallocated Destination Number Caused Treatment*

Note that the **ANSI** field in the **Reason** header is an extension to *RFC 3326* and *draft-jesske-dispatch-reason-in-responses-01*. This field is used in the same fashion as **Q.850** fields, with the exception that the cause values correspond to **ANSI** cause values defined in *ANSI T1.113/2000*.

When a non-dipped or dipped-non-ported number (*telephone-subscriber* portion of the **Request-URI**) is within a number block allocated to the customer but has no subscriber assigned, the *C-SCP* platform should respond with a 404 Not Found response with **Reason** headers as shown in [Message 6.19](#).

```

SIP/2.0 404 Not Found
Via: SIP/2.0/UDP cp.heraamax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heraamax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heraamax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=1 ;text="unallocated (unassigned) number"
Content-Length: 0
Message 6.19: Unallocated Destination Number Caused Treatment

```

Note that the **ANSI** field in the **Reason** header is an extension to *RFC 3326* and *draft-jesske-dispatch-reason-in-responses-01*. This field is used in the same fashion as Q.850 fields, with the exception that the cause values correspond to ANSI cause values defined in *ANSI T1.113/2000*.

No route to destination

Send special information tone

Misdialled trunk prefix

User busy

Subscriber absent

Call rejected

Number changed

If the number (*telephone-subscriber* portion of the **Request-URI**) has been changed and redirect is in force, the *C-SCP* should respond with a 404 Not Found response with **Reason** headers as shown in [Message 6.20](#).

```

SIP/2.0 404 Not Found
Via: SIP/2.0/UDP cp.heraamax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heraamax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heraamax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=22 ;text="number changed"
Content-Length: 0
Message 6.20: Number Changed Caused Treatment - 404 Not Found Response

```

or, alternately if automatic redirect is to be performed, a 301 Moved Permanently response with **Reason** headers as shown in [Message 6.21](#).

```
SIP/2.0 301 Moved Permanently
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <tel:+19998887777>
Reason: Q.850 ;cause=22 ;text="number changed"
Content-Length: 0
Message 6.21: Number Changed Caused Treatment – 301 Moved Permanently Response
```

where ‘999-888-7777’ is the new number.

Redirect to new destination

Destination out of order

Invalid number format (address incomplete)

If the destination number format (*telephone-subscriber* portion of the **Request-URI**) is invalid (e.g. the number is an international rather than national number), the *C-SCP* should respond with a **484 Address Incomplete** response with **Reason** headers as shown in [Message 6.22](#).

```
SIP/2.0 484 Address Incomplete
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=28 ;text="invalid number format (address incomplete)"
Content-Length: 0
Message 6.22: Invalid Number Format Caused Treatment
```

Normal, unspecified

No circuit/channel available

Network out of order

Should the network be out of order the *C-SCP* platform should return a **500 Server Internal Error** response with **Reason** headers as shown in [Message 6.23](#).

```

SIP/2.0 500 Server Internal Error
Via: SIP/2.0/UDP cp.heraamax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heraamax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heraamax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=38 ;text="network out of order"
Content-Length: 0

```

Message 6.23: *Network Out-of-order Caused Treatment*

Temporary failure

Should a temporary failure occur in the *C-SCP* platform, either associated with the platform or this specific dialog, it should return a 500 Server Internal Error response with Reason headers as shown in [Message 6.24](#).

```

SIP/2.0 500 Server Internal Error
Via: SIP/2.0/UDP cp.heraamax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heraamax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heraamax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=41 ;text="temporary failure"
Content-Length: 0

```

Message 6.24: *Temporary Failure Caused Treatment*

Switching equipment congestion

Should the *Tel-Lingu* platform be experiencing congestion, it should return a 500 Server Internal Error response with Reason headers as shown in [Message 6.25](#).

```

SIP/2.0 500 Server Internal Error
Via: SIP/2.0/UDP cp.heraamax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heraamax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heraamax.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=42 ;text="switching equipment congestion"
Content-Length: 0

```

Message 6.25: *Switching Equipment Congestion Caused Treatment*

Resource unavailable, unspecified

Misrouted call to a ported number

When a dipped ported number is not found (that is, there is a *npdi* and *rn* parameter in the Request-URI, but the *telephone-subscriber* portion of the Request-URI is not allocated to a customer) the *C-SCP* platform should send a 404 Not Found response with Reason headers as shown in [Message 6.26](#).


```

SIP/2.0 404 Not Found
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Reason: ANSI ;cause=26 ;text="misrouted call to a ported number"
Reason: Q.805 ;cause=14 ;text="QoR: ported number"
Content-Length: 0
Message 6.26: Misrouted Call to Ported Number Caused Treatment

```

Note that the ANSI field in the Reason header is an extension to *RFC 3326* and *draft-jesske-dispatch-reason-in-responses-01*. This field is used in the same fashion as Q.850 fields, with the exception that the cause values correspond to ANSI cause values defined in *ANSI T1.113/2000*.

Number portability Query on Release – number not found

When a non-dipped number (*telephone-subscriber* portion of the Request-URI with no *npdi* parameter in the Request-URI) is found to be ported out, the response should be a 404 Not Found response with Reason headers as shown in [Message 6.27](#).

```

SIP/2.0 404 Not Found
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=182.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Reason: ANSI ;cause=27 ;text="NP QoR - number not found"
Reason: Q.850 ;cause=14 ;text="QoR: ported number"
Content-Length: 0
Message 6.27: Number Not Found Caused Treatment

```

6.2.3.2 Example of an Unsuccessful Call from PSTN to SIP

Refer to [Figure 6.1](#) for the messages exchanged that correspond with the numbers below.

1. Point (1) in [Figure 6.1](#).

A call is generated by the PSTN and an IAM is sent to the *Optranex 248* as the terminating switch. The IAM for this example contains the information shown in [Example 6.4](#).

<i>FCI</i>	number translated
<i>CgPN</i>	NSN, 222-333-4444, presentation-restricted
<i>CdPN</i>	NSN, 777-666-5555
<i>JIP</i>	830-660
<i>GAP</i>	ported number, NSN, 999-666-5555
<i>GAP</i>	dialed number, NSN, 444-444-4444, presentation-allowed
<i>GAP</i>	additional user provided number - not verified, NSN, 666-666-6666, presentation-allowed
<i>GN</i>	calling name, name available/unknown, "Joe", presentation-allowed
<i>GN</i>	original called name, name available/unknown, "Sue", presentation-allowed
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0

Example 6.4: *IAM received for Unsuccessful PSTN to SIP Call Example*

For the purpose of this example, the LRN present in the CdPN (777-666-5555) is a correct LRN for the *Optranex 248*, however, the ported number (999-666-5555) is no longer ported here and has been ported out.

2. Point (2) in [Figure 6.1](#).

The *Optranex 248* translates the incoming IAM to a SIP INVITE and sends it to the *Carrier Proxy* for processing.

```
INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001
From: "Joe" <tel:+16666666666>;tag=XXX
To: "Sue" <tel:+14444444444>
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+16666666666
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com
@gw1.heramax.com:5060;user=phone;transport=udp>
Max-Forwards: 33
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Content-Type: application/sdp
Content-Length: 134
```

```
v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Message 6.28: *Optranex 248 INVITE for Unsuccessful PSTN to SIP Call Example*

Notes:

- The **tel** URI in the Request-URI contains the termination number. In this case, the

number is ported and was translated as indicated by the *npdi* parameter, and the LRN associated with the ported number is provided in the *rn* parameter.

- The **From** and **To** header contain the additional user provided calling number and the dialed number from the IAM.
- The **Contact** header provides only displayable information about the originator.
- The **Max-Forwards** header value is calculated by applying a weighting factor to the received *ISUP Hop Counter* parameter.
- The **P-Asserted-Identity** header provides private information about the originator, including the calling party number, calling party's category, originating line information, jurisdiction information parameter, incoming trunk group and terminating gateway switch.
- the **P-Charge-Info** header provides the private information about the originator, including the charge number, calling party's category, originating line information, jurisdiction information parameter, incoming trunk group and terminating gateway switch.
- The *trunk-context* parameter, in the various headers in which it appears, contains the hostname associated with the specific *Optranex 248* gateway switch that received the call. The *tgrp* parameter is only meaningful within the context of this gateway switch.

3. Point (3) in [Figure 6.1](#).

The *Carrier Proxy* forwards the INVITE to the *C-SCP* for further processing.

```
INVITE tel:+19996665555;npdi;rn=+17776665555 SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z8hG4bK-AAAAAA
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XXX
To: "Sue" <tel:+14444444444>
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+16666666666
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com
@gw1.heramax.com:5060;user=phone;transport=udp>
Max-Forwards: 32
P-Asserted-Identity: Joe
<tel:+12223334444;cpc=ordinary;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Privacy: id
P-Charge-Info: <tel:+12223334444;oli=00;rn=+1830660
;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Message 6.29: Carrier Proxy INVITE for Unsuccessful PSTN to SIP Call Example
```

Note that the INVITE message forwarded is not altered from the INVITE message received with the exception of the header fields which must be altered for proxy behaviour in accordance with *RFC 3261*.

4. Point (4) in [Figure 6.1](#).

The *Carrier Proxy*, because it has forwarded the INVITE message and cannot determine when it will receive a response, responds to the incoming invite with a 100 Trying message.

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Content-Length: 0
```

Message 6.30: *Carrier Proxy 100 Trying for Unsuccessful PSTN to SIP Call Example*

5. Point (5) in [Figure 6.1](#).

The *C-SCP* platform determines that it will respond in sufficient time and skips sending a 100 Trying message.

6. Point (6) in [Figure 6.1](#).

The *C-SCP* looks up the destination number (999-666-5555) and determines that the number does not belong to a customer. Because the *Request-URI* contains a *npdi* indication, as well as a *rn* routing number, it surmizes that the call to the ported number has been misrouted. So, it responds with a 404 Not Found response and includes the *Reason* header with cause values appropriate for a misrouted call to a ported number.

```
SIP/2.0 404 Not Found
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z8hG4bK-AAAAAA;received=192.168.9.30
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Reason: ANSI ;cause=26 ;text="misrouted call to a ported number"
Reason: Q.805 ;cause=14 ;text="QoR: ported number"
Content-Length: 0
```

Message 6.31: *C-SCP 404 Not Found Response for Unsuccessful PSTN to SIP Call Example*

7. Point (7) in the [Figure 6.1](#).

The *Carrier Proxy*, upon receiving the 4xx or 5xx response, forwards the response to the *Optranex 248* gateway switch.

```
SIP/2.0 404 Not Found
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 INVITE
Reason: ANSI ;cause=26 ;text="misrouted call to a ported number"
Reason: Q.805 ;cause=14 ;text="QoR: ported number"
Content-Length: 0

Message 6.32: Carrier Proxy 404 Not Found Response for Unsuccessful PSTN to SIP Call Example
```

8. Point (8) in the [Figure 6.1](#).

Upon receiving the 404 Not Found response, the *Optranex 248* switch acknowledges receipt of the 4xx message.

```
ACK sip:anonymous@gw1.heramax.com SIP/2.0
Via: SIP/2.0/UDP gw1.heraax.com:5060;branch=z9hG4bK-000001
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 ACK
Content-Length: 0

Message 6.33: Optranex 248 ACK for Unsuccessful PSTN to SIP Call Example
```

9. Point (9) in the [Figure 6.1](#).

The *Carrier Proxy*, upon receiving the ACK from the *Optranex 248*, forwards the request to the *C-SCP*.

```
ACK sip:anonymous@gw1.heramax.com SIP/2.0
Via: SIP/2.0/UDP cp.heraax.com:5060;branch=z9hG4bK-AAAAAA
Via: SIP/2.0/UDP gw1.heraax.com:5060;branch=z9hG4bK-000001;received=192.168.9.10
From: "Joe" <tel:+16666666666>;tag=XX
To: "Sue" <tel:+14444444444>;tag=YY
Call-ID: 111222@gw1.heramax.com
CSeq: 1 ACK
Content-Length: 0

Message 6.34: Carrier Proxy ACK for Unsuccessful PSTN to SIP Call Example
```

10. Point (10) in the [Figure 6.1](#).

While acknowledging the receipt of the 404 Not Found response, the *Optranex 248* switch formulates an ISUP REL message and includes cause value 26 ("misrouted call to a ported number") and releases the call setup.

Note that the release cause is taken directly from the **Reason** headers of the received 404 Not Found response. The ANSI release cause is used in preference to any Q.850 release cause present. When no release cause is present (neither **Reason** header field is present), the release cause indicating failure beyond an interworking point will be used. Also note that the location of the cause will be beyond and interworking point.

11. Point (11) in the [Figure 6.1](#).

The originating PSTN switch completes the release of the call by sending a RLC message to the *Optranex 248* switch.

6.3 Calls from SIP to PSTN

Calls from SIP (IMS CN) to the PSTN (CS Domain) are illustrated in [Figure 6.3](#).

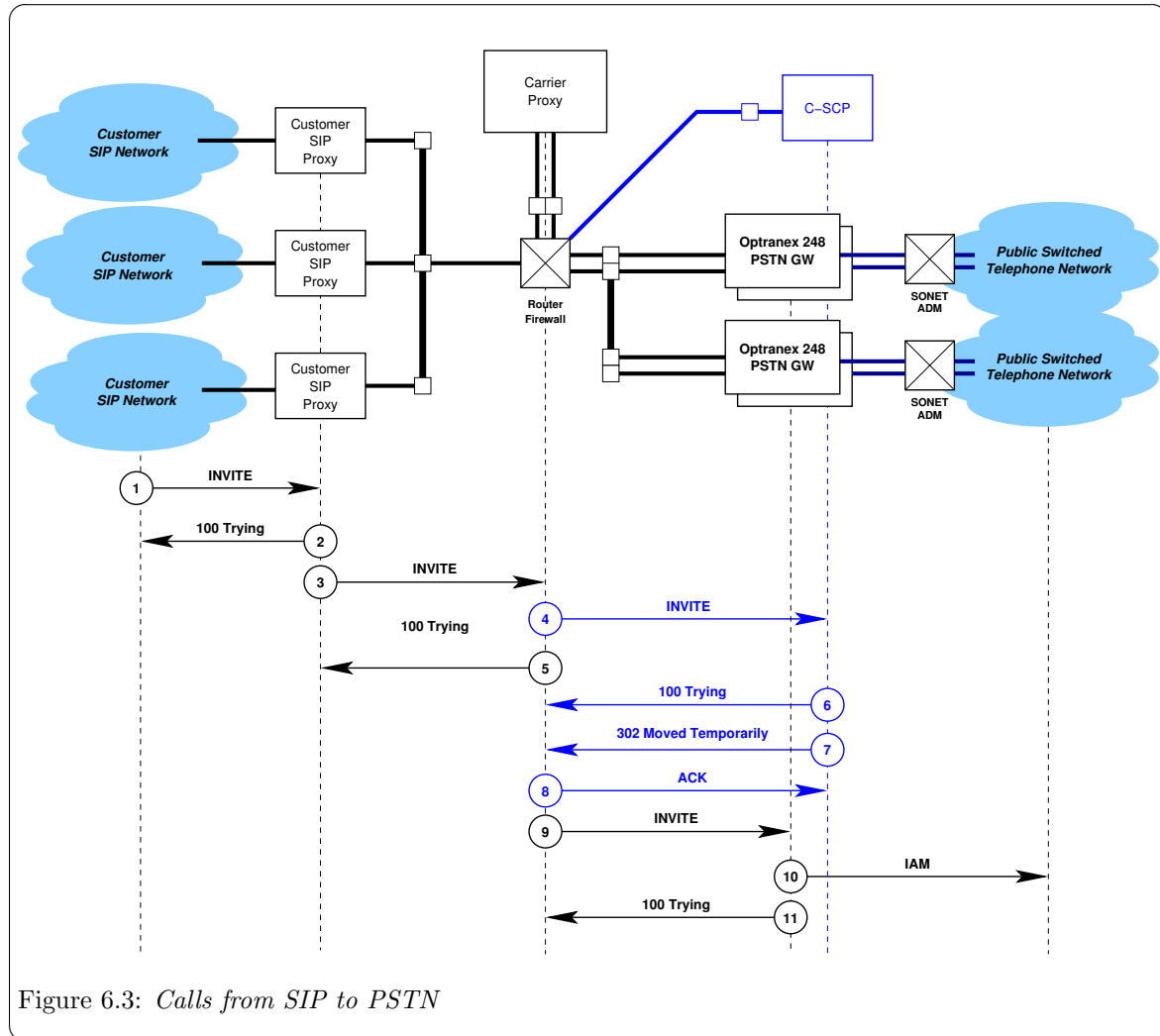


Figure 6.3: *Calls from SIP to PSTN*

Call flows shown in blue (between the *Carrier Proxy* and the *C-SCP*) are now transparent to both the *Customer Proxy* and the *Optranex 248* gateway switch and are no longer specified by this document (neither the *Customer Proxy* nor the *Optranex 248* gateway switch need to know that the *C-SCP* exists).

1. Point (1) in [Figure 6.3](#).

A user agent (UA) in a customer SIP network initiates a call and launches an INVITE message to the customer's outgoing SIP proxy. An example of an INVITE message is shown in [Message 6.35](#).

```

INVITE sip:+19725552222@ss1.customer1.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
;tag=0fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Contact: <sip:alice@client.customer1.com;transport=udp>
Proxy-Authorization: Digest username="alice", realm="customer1.com",
nonce="dc3a5ab25302aa931904ba7d88fa1cf5", opaque="",
uri="sip:+19725552222@ss1.customer1.com;user=phone",
response="ccdca50cb091d587421457305d097458c"
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.customer1.com
s=-
c=IN IP4 client.customer1.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Message 6.35: UA INVITE for Successful SIP to PSTN Call

```

2. Point (2) in [Figure 6.3](#).

The customer's outgoing SIP proxy acknowledges receipt of the INVITE with a 100 Trying response. An example of the response is shown in [Message 6.36](#).

```

SIP/2.0 100 Trying
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Content-Length: 0

Message 6.36: Customer Proxy 100 Trying Response for Successful SIP to PSTN Call

```

3. Point (3) in [Figure 6.3](#).

The customer's outgoing SIP proxy determines that the call is to be sent to the PSTN and forwards the INVITE message to the *Carrier Proxy*. In performing this function, the customer's outgoing SIP proxy is performing the *I-CSCF* (Interrogating Call Services Control Function) of the 3GPP IMS architecture and the *Carrier Proxy* is acting in the role of the *BGCF* (Breakout Gateway Control Function).


```

INVITE sip:+19725552222@cp.heramax.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.customer1.com;lr>
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Contact: <sip:alice@client.customer1.com;transport=udp>
Content-Type: application/sdp
Content-Length: 154

```

```

v=0
o=alice 2890844526 2890844526 IN IP4 client.customer1.com
s=-
c=IN IP4 client.customer1.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.37: *Customer Proxy INVITE for Successful SIP to PSTN Call*

4. Point (4) in [Figure 6.3](#).

The *Carrier Proxy* forwards the INVITE message received from the customer's outgoing SIP proxy to the *C-SCP* platform for authorization and route selection. Calls coming from the SIP network will have the INVITE from the customer SIP proxy forwarded directly to the *C-SCP* platform without the need for additional processing at the *Carrier Proxy*.

When the customer's outgoing SIP proxy is not considered part of the *Carrier Proxy*'s trust network, the *Carrier Proxy* strips all trust headers from the INVITE before forwarding the information on to the *C-SCP*. When the customer's outgoing SIP proxy is considered part of the *Carrier Proxy*'s trust network, the *Carrier Proxy* proxies the INVITE message to the *C-SCP* (without modifying any non-proxy headers or fields).

```

INVITE sip:+19725552222@telingua.heramax.com;user=phone SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 68
Record-Route: <sip:ss1.customer1.com;lr>
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Contact: <sip:alice@client.customer1.com;transport=udp>
Content-Type: application/sdp
Content-Length: 154

```

```

v=0
o=alice 2890844526 2890844526 IN IP4 client.customer1.com
s=-
c=IN IP4 client.customer1.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.38: *Carrier Proxy C-SCP INVITE for Successful SIP to PSTN Call*

5. Point (5) in [Figure 6.3](#).

The *Carrier Proxy* optionally (at some point when it determines that a retransmission of an INVITE might occur before a response is issued, not necessarily at the point shown) issues a 100 Trying message to the customer proxy.

```

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Content-Length: 0

```

Message 6.39: *Optranex 248 100 Trying Response for Successful SIP to PSTN Call*

6. Point (6) in [Figure 6.3](#).

The *C-SCP* platform acknowledges receipt of the INVITE with a 100 Trying response.

```

SIP/2.0 100 Trying
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.30
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Content-Length: 0

Message 6.40: C-SCP 100 Trying Response for Successful SIP to PSTN Call

```

7. Point (7) in Figure 6.3.

The *C-SCP* platform determines that the client is authorized to connect to the PSTN and populates route information in the **Contact** headers of a **302 Moved Temporarily** response. In performing this function the *C-SCP* platform performs the role of a *BGCF* (Breakout Gateway Control Function) in the *3GPP TS 23.002* architecture.

The *C-SCP* platform must implement the dial plan. I would suggest that the *Sansay* proxies and customer SIP phones be configured to send *tel* URL formatted numbers or *tel* URL formatted numbers in a **sip** URI¹ in the **To**, **From** and the **Request-URI** fields. Otherwise the *C-SCP* platform must be able to accept any of the possible URI variations.²

Should the *C-SCP* platform discover a difficulty with the **addr-spec** in a *sip* URI or **telephone-subscriber** in a *tel* URL, the platform can respond to the INVITE with a normal 4xx or 5xx response which will be relayed back to the customer proxy.

To direct the call to the telephone network, the *C-SCP* should formulate a **302 Moved Temporarily** response and place the destination number parameters and route information in the **Contact** headers present in the message.

The **Contact** header should be formatted as a SIP URI so that the identity of the selected gateway can be provided to the *Carrier Proxy*.

For example:

¹ For formatting of **tel** URLs in **sip** URIs, see RFC 3261 section 19.1.6.

² See RFC 4504 Section 5.3 for variations on addressing provided by a SIP phone.

```

SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.30
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Contact: Bob <sip:+19725552222;npdi
      ;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com@gw1.heramax.com>
      ;user=phone
Contact: Bob <sip:+19725552222;npdi
      ;tgrp=EDTNAB01DS1;trunk-context=gw2.heramax.com@gw2.heramax.com>
      ;user=phone
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
P-Asserted-Identity: Alice <tel:+13145551111;cpc=ordinary;oli=00;rn=+1830660>
Privacy: none
CSeq: 1 INVITE
Content-Length: 0

Message 6.41: C-SCP 302 Moved Temporarily Response for Successful SIP to PSTN Call

```

Notes:

- The **Contact** headers in the 302 Moved Temporarily response must be formatted as a **sip** URI (so that the resulting INVITE message will be forwarded to the correct *Optranex 248* gateway switch).
 - The **Contact** headers in the 302 Moved Temporarily response must contain the dip indicator, *npdi*, and, when the number is a ported number, must contain the LRN in the *rn* parameter.
 - Each **Contact** header in the 302 Moved Temporarily response must contain *trunk-context* and *tgrp* parameters indicating the domain name of the specific *Optranex 248* gateway switch selected and the trunk group (route) to use on that switch. The domain name portion of the **sip** URI must also contain the domain name of the same *Optranex 248* gateway switch.
 - The **P-Asserted-Identity** header, if any, in the 302 Moved Temporarily response must be formatted as a **tel** URI.
 - A **P-Asserted-Identity** header must be included if the calling party number or calling party's category is to be included in the resulting IAM.
 - The **P-Charge-Info** header, if any, in the 302 Moved Temporarily response must be formatted as a **tel** URI.
 - A **P-Charge-Info** header must be included if the charge number, originating line information, or jurisdiction information parameter is to be included in the resulting IAM.
 - Any **Privacy** header included in the 302 Moved Temporarily response will be used to control the presentation of the calling party number and calling name (ISUP CNAM method). A setting of *id* or higher protects the calling party number; *user* or higher protects the calling name.
8. Point (8) in [Figure 6.3](#).

The *Carrier Proxy* acknowledges receipt of the 302 Moved Temporarily response with an ACK message.

```

ACK sip:tellingua@customer1.com SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 70
Route: <sip:gw1.heramax.com;lr>
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
      ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 ACK

```

Message 6.42: *Carrier Proxy ACK for Successful SIP to PSTN Call*

9. Point (9) in [Figure 6.3](#).

The *Carrier Proxy* adds the contents of the *Contact* header into its target set and issues an *INVITE* message to the first target.

```

INVITE sip:+19725552222;npdi;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com\
@gw1.heramax.com;user=phone SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 67
Record-Route: <sip:ss1.customer1.com;lr>
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Contact: <sip:alice@client.customer1.com;transport=udp>
P-Asserted-Identity: Alice <tel:+13145551111;cpc=ordinary;oli=00;rn=+1830660>
Privacy: none
Content-Type: application/sdp
Content-Length: 154

```

```

v=0
o=alice 2890844526 2890844526 IN IP4 client.customer1.com
s=-
c=IN IP4 client.customer1.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

```

Message 6.43: *Carrier Proxy retargetted INVITE for Success SIP to PSTN Call*

Notes:

- The *Request-URI* contains the **sip** URI exactly as returned in the 302 Moved Temporarily response *Contact* header in accordance with *RFC 3261*.
- The *P-Asserted-Identity*, *P-Charge-Info* and *Privacy* headers are taken directly from the 302 Moved Temporarily response. This is consistent with proxy behaviour.

10. Point (10) in [Figure 6.3](#).

The *Optranex 248* translates the incoming INVITE message and the headers to formulate an IAM which is sent to the PSTN.

The INVITE and headers received by the *Optranex 248* platform is mapped to an IAM message as detailed in [\[\[\\(undefined\\)\]\(#\)\], page \[\\(undefined\\)\]\(#\)](#). This process results in the IAM shown in [Example 6.5](#), which is sent to the SS7 network.

<i>FCI</i>	number translated
<i>CgPN</i>	NSN, 314-555-1111, presentation-allowed
<i>CdPN</i>	NSN, 972-555-2222
<i>JIP</i>	830-660
<i>GN</i>	calling name, name available/unknown, "Alice", presentation-allowed
<i>GN</i>	original called name, name available/unknown, "Bob", presentation-allowed
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0

Example 6.5: IAM sent for Successful SIP to PSTN Call

After this point, the *Carrier Proxy* platform is no longer necessarily in the message exchange which follows along according to *RFC 3261*, *RFC 3666*, *RFC 3372*, *ITU-T Q.1912.5* and *3GPP TS 29.163*.³

6.3.1 Translation of INVITE to IAM

Fields from the received INVITE are mapped to the outgoing IAM parameter by the *Optranex 248* as follows:

- If the *trunk-context* parameter in the *Request-URI* does not correspond to the receiving *Optranex 248* platform, a 502 **Bad Gateway** response will be returned and no IAM will be generated.
- The translated IAM will be sent on the trunk group specified with the *tgrp* parameter of the *Request-URI*. If all trunks are busy in the trunk group, a 503 **Service Unavailable** response will be generated so that the *Carrier Proxy* can try the next route. When all routes are busy, the *Carrier Proxy* will issue treatment toward the SIP proxy (e.g. 503 **Service Unavailable**).
- If the *Request-URI* header has *npdi* parameter set in the *tel* URI formatted *user-info* portion of the *Request-URI* in accordance with *RFC 4694*, the *number translated* bit will be set in the *Forward Call Indicators* of the IAM.
- The *telephone-subscriber* from the *tel* URI of the *Request-URI* (or if a *rn* parameter is present, the contents of the *rn* parameter) will be used to populate the *Called Party Number* of the IAM. This is consistent with the behavior of the *MGCF* described in *3GPP TS 29.163 7.2.3.1.2A* and *RFC 4694*.
- If the *rn* parameter exists in the *Request-URI user-info tel* URI portion, the *telephone-subscriber* part of the *tel* URI will be used to populate the ported number in a *Generic Address Parameter* in the IAM. This is consistent with the behavior of the *MGCF* described in *3GPP TS 29.163 7.2.3.1.2A* and *RFC 4694*.
- If the *cic* parameter exists in the *Request-URI tel* URI in accordance with *RFC 4694*, the value of this parameter will be used to populate a *Carrier Identification* parameter in the outgoing IAM.

³ Note that the IP address and UDP port number in the 180 **Ringling** or OK response to the original INVITE is mapped from the bearer circuit selected for the call.

- If the *dai* parameter exists in the Request-URI tel URI in accordance with *draft-yu-tel-dai-08*, the value of this parameter will be used to populate the *Carrier Selection Information* parameter in the outgoing IAM.
- If there is a recognized *telephone-subscriber* portion in the To header and it does not correspond to the *Called Party Number*, a "dialled number" *Generic Address Parameter* will be added to the outgoing IAM.
- If there is a *display-name* portion in the To header and the ISUP CNAM method is marked on the trunk group, a "original called name" *Generic Name* parameter that is *presentation-allowed* will be added to the outgoing IAM.
- If a *cpc* parameter exists in the P-Asserted-Identity header, a *Calling Party Category* parameter will be included in the IAM with the value indicated. This behaviour is consistent with that described for the MGCF in *3GPP TS 29.163 Annex C.1*.⁴
- If a *oli* parameter exists in the P-Charge-Info or P-Asserted-Identity header; an *Originating Line Information* parameter will be included in the IAM with the value indicated. This behaviour is consistent with that described for the MGCF in *3GPP TS 29.163 Annex H.1*.⁵
- If an *rn* parameter exists in the P-Charge-Info or P-Asserted-Identity header, it will be used to populate the *Jurisdiction Information Parameter* in the outgoing IAM. This behaviour is consistent with *RFC 4694*.⁶
- If a valid *telephone-subscriber* number is present in the P-Charge-Info header, and it is different from the *Calling Party Number* when the *noa* parm is 1, 2, 3 or absent, or, it is different from the *Called Party Number* when the *noa* parm is 5, 6 or 7, it will be used in the *Charge Number* parameter in the outgoing IAM. When a *npi* or *noa* parameter is also present in the P-Charge-Info header, the numbering-plan-indicator and nature-of-address in the *Charge Number* parameter will be populated accordingly. This behaviour is consistent with that described in *draft-york-sipping-p-charge-info-07*.
- If a valid *telephone-subscriber* number is present in the P-Asserted-Identity header, it will be used in the *Calling Party Number* parameter of the outgoing IAM. If a *Privacy* header exists and is set to 'id' (or wider scope), the *Calling Party Number* will be marked *presentation-restricted*; otherwise, it will be marked *presentation-allowed*.
- If a *display-name* exists in the P-Asserted-Identity header and the trunk is configured for ISUP CNAM method, the *display-name* will be placed in a calling name *Generic Name* parameter in the outgoing IAM. If a *Privacy* header exists and is set to 'user' (or wider scope), the *Generic Name* will be marked *presentation-restricted*; otherwise, it will be marked, *presentation-allowed*.
- If there is a recognized *telephone-subscriber* portion in the From header, and it is different from that of the P-Asserted-Identity header, an "additional user provided number - not verified" *Generic Address Parameter* will be placed in the outgoing IAM with *presentation-allowed*.
- If there is a *display-name* portion in the From header, and the outgoing trunk is marked for ISUP CNAM, a "calling name" *Generic Name* parameter will be placed in the outgoing IAM marked *presentation-allowed* (if one is not already present).

⁴ See *3GPP TS 29.163 V9.0.0 (2009-12) Annex C.1* and *draft-patel-dispatch-cpc-oli-parameter-02*.

⁵ See *3GPP TS 29.163 V9.0.0 (2009-12) Annex H.1* and *draft-patel-dispatch-cpc-oli-parameter-02*.

⁶ See *RFC 4694* Section 5.2.4. Note, however, that neither *ITU-T Q.1912.5/1998* nor *3GPP TS 29.163 V9.0.0* support the concept of communicating the *Jurisdiction Information Parameter* from the INVITE to the IAM.

6.3.1.1 Example of INVITE to IAM Translation

So, for the SIP INVITE message last presented, and repeated in [Message 6.44](#),

```
INVITE sip:+19725552222;npdi;tgrp=EDTNAB01DS1;trunk-context=gw1.heramax.com\
@gw1.heramax.com;user=phone SIP/2.0
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 67
Record-Route: <sip:ss1.customer1.com;lr>
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Contact: <sip:alice@client.customer1.com;transport=udp>
P-Asserted-Identity: Alice <tel:+13145551111;cpc=ordinary;oli=00;rn=+1830660>
Privacy: none
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.customer1.com
s=-
c=IN IP4 client.customer1.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Message 6.44: 302 Moved Temporarily Response for SIP to PSTN Call Example
```

the outgoing IAM would be as shown in [Example 6.6](#).

<i>FCI</i>	number translated
<i>CdPN</i>	NSN, 999-888-7777
<i>CgPN</i>	NSN, 222-333-4444, presentation-restricted
<i>GAP</i>	ported number, NSN, 999-666-5555
<i>GAP</i>	dialed number, NSN, 444-444-4444, presentation-allowed
<i>GAP</i>	additional user provided number - not verified, NSN, 666-666-6666, presentation-allowed
<i>JIP</i>	830-660
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0
<i>GN</i>	calling name, "Joe", presentation-allowed
<i>GN</i>	called name, "Sue", presentation-allowed

Example 6.6: *IAM sent for Successful SIP to PSTN Call Example*

6.3.2 Processing of Calls Terminating on the PSTN

For calls terminating on the PSTN, the *Carrier Proxy* is responsible for controlling interactions with the *C-SCP*. In conjunction with the *C-SCP*, the *Carrier Proxy* is responsible for formatting a

SIP INVITE message toward the *Optranex 248* that contains information appropriate for the SIP to ISUP interworking.

The functions of the *Carrier Proxy* in its interaction with the *Customer Proxy* and *Optranex 248* does not need to deviate from the basic stateless or stateful transaction behaviour described in *RFC 3261*. As such, any popular commercial or open-source implementation of a SIP proxy server can be used to perform this function. Nevertheless, the specific interaction between the *Carrier Proxy* and the *C-SCP* is implementation and network specific and does not really need to conform to a standard.

The steps involved are as follows:

- Forwarding the initial SIP INVITE, received by the *Carrier Proxy* to the *C-SCP* platform. This is detailed in [\[undefined\] \[undefined\], page undefined](#).
- Extracting information from the forwarded SIP INVITE at the *C-SCP* platform. This is detailed in [\[undefined\] \[undefined\], page undefined](#).
- Generating a 302 Moved Temporarily response at the *C-SCP* and processing the response at the *Carrier Proxy*. This is detailed in [\[undefined\] \[undefined\], page undefined](#).
- Generating the SIP INVITE message sent to the *Optranex 248* gateway switch at the *Carrier Proxy*. This is detailed in [\[undefined\] \[undefined\], page undefined](#).

A complete example is provided in [\[undefined\] \[undefined\], page undefined](#).

6.3.2.1 Forwarding the Initial SIP INVITE

When the *Carrier Proxy* receives a SIP INVITE message from a *Customer Proxy*, it determines whether the call is to be forwarded to the PSTN, on whether some other action is to be performed on the nascent session. This determination could include querying a Carrier ENUM database, or some other mechanism. Upon deciding that the calls is to be forwarded to the PSTN through a *Optranex 248* gateway switch, the *Carrier Proxy* processing the message and forwards a SIP INVITE message to the *C-SCP*. In processing the received INVITE message and formulating a SIP INVITE message to send to the *C-SCP*, the *Carrier Proxy* should perform all of the actions necessary for trust domains, particularly with regards to P-Requested-Identity, P-Asserted-Identity, P-Charge-Info and Privacy fields. That is, any untrusted private headers received from the *Customer Proxy* should be stripped unless the customer proxy is within the same trust domain as the *Carrier Proxy*. Examples are provided in [\[Example of a Successful Call from SIP to PSTN\], page 78](#), for Point (3) and (4) in [Figure 6.3](#).

6.3.2.2 Extracting Information from the SIP INVITE

Examples are provided in [\[Example of a Successful Call from SIP to PSTN\], page 78](#), for Point (5) and (6) in [Figure 6.3](#).

6.3.2.3 302-Moved-Temporarily Response

Formulating the 302-Moved-Temporarily Response

Examples are provided in [\[Example of a Successful Call from SIP to PSTN\], page 78](#), for Point (6) and (7) in [Figure 6.3](#).

Processing the 302-Moved-Temporarily Response

Examples are provided in [\[Example of a Successful Call from SIP to PSTN\], page 79](#), for Point (8) and (9) in [Figure 6.3](#).

6.3.3 Generating the Gateway INVITE

The *Carrier Proxy* is responsible for analyzing information from the received **302 Moved Temporarily** response, incorporating that information into the target set for the session and formulating a SIP **INVITE** message to the first of the indicated gateways (as indicated by the **Contact** headers in the **302 Moved Temporarily** response). When formulating the **INVITE** message, the *Carrier Proxy* should take the actions described in *RFC 3261* with the following additional considerations:

Examples are provided in [Example of a Successful Call from SIP to PSTN], page 79, for Point (9) in Figure 6.3.

- The *Carrier Proxy* should not return the **302 Moved Temporarily** response message received from the *C-SCP* back to the *Customer Proxy* for processing, but must process the response itself.
- The *Carrier Proxy* must not fork the target set resulting from the **302 Moved Temporarily** response message.
- Any **P-Asserted-Identity**, **Privacy**, and **P-Charge-Info** headers present in the **302 Moved Temporarily** response message or to be added by the *Carrier Proxy* should be forwarded to the *Optranex 248* platform.

6.3.3.1 Example of a Successful Call from SIP to PSTN

1. Point (1) in Figure 6.3.

A SIP **INVITE** message is launched by a UA in the customer SIP network initiating a session with the *Customer Proxy*.

The precise format of this message is not specified by this document, but, however, is envisioned to be similar to the following:

2. Point (2) in Figure 6.3.

The *Customer Proxy* optionally responds with a **100 Trying** message.

3. Point (3) in Figure 6.3.

The *Customer Proxy* determines that the session is to be established within the **heramax.com** network and forwards the message to the *Carrier Proxy*. In forwarding the **INVITE** to the *Carrier Proxy*, the *Customer Proxy* should utilize DNS **NAPTR** and **SRV** entries to provide redundancy between images of the *Carrier Proxy*. Also note that the *Carrier Proxy* can easily support UDP, TCP and SCTP, both with and without TLS by utilizing software such as *SER* or *OpenSIPS*.⁷

4. Point (4) in Figure 6.3.

The *Carrier Proxy* determines that the call is destined toward the PSTN and forwards a header-filtered **INVITE** message to the *C-SCP*.

The precise format of this message is no longer specified by this document, but, however, was envisioned to be similar to the following:

5. Point (5) in Figure 6.3.

The *Carrier Proxy* optionally responds with a **100 Trying** message to suppress retransmission of the **INVITE**.

6. Point (6) in Figure 6.3.

The *C-SCP* optionally responds with a **100 Trying** message to suppress retransmission of the **INVITE**.

⁷ *OpenSIPS* was formerly called *OpenSER*.

7. Point (7) in [Figure 6.3](#).

The *C-SCP* processes the received INVITE and formulates a 302 Moved Temporarily response.

The precise format of this message is no longer specified by this document, but, however, was envisioned to be similar to the following:

8. Point (8) in [Figure 6.3](#).

The *Carrier Proxy* acknowledges the 302 Moved Temporarily response.

9. Point (9) in [Figure 6.3](#).

The *Carrier Proxy* formulates a branch INVITE message from the original INVITE message received from the *Customer Proxy* and the 302 Moved Temporarily response formulated by the *C-SCP* and sends the INVITE message to the appropriate target within the current target set defined by the 302 Moved Temporarily response message.

10. Point (10) in [Figure 6.3](#).

The selected *Optranex 248* gateway switch receives the INVITE message from the *Carrier Proxy* and generates a corresponding ISUP IAM message which is sent to the SS7 network.

11. Point (11) in [Figure 6.3](#).

The *Optranex 248* optionally response with a 100 Trying message.

12. Remainder of Session

The remainder of the session establishment is according to normal SIP to ISUP conversion mechanisms (*SIP-T*, *Q.1912.5*, *3GPP TS 29.163*). The *Carrier Proxy* can decide at the time that it issues the INVITE message to the *Optranex 248* platform whether it wishes to continue receiving requests or not.

6.3.4 Treatment of SIP Originating Calls

6.3.4.1 Treatment of SIP Calls

Q.850 Cause Values

The mapping of *Q.850* cause values returned in an ACM or REL message to SIP responses is illustrated: for the normal class of cause values, in [Table 6.4](#); resource unavailable class, [Table 6.5](#); service or option not available class, [Table 6.6](#).⁸ These *Q.850* cause values will be mapped to the corresponding SIP response listed in the tables. SIP 4xx or 5xx response, CANCEL or BYE message will include the Reason header that contains the *Q.850* cause value. An example is provided in [Message 6.45](#).

⁸ Note that this mapping is consistent with *Table 21/Q.1912.5*.

Q.850	Description	Q.1912.5 SIP Response	RFC3398
1	<i>unallocated (unassigned) number</i>	404 Not Found	404
2	<i>no route to network</i>	500 Server Internal Error	404
3	<i>no route to destination</i>	500 Server Internal Error	404
4	<i>send special information tone</i>	500 Server Internal Error	–
5	<i>misdialled trunk prefix</i>	404 Not Found	–
8	<i>preemption</i>	500 Server Internal Error	–
9	<i>preemption-circuit reserved for reuse</i>	500 Server Internal Error	–
16	<i>normal call clearing</i>	–	–
17	<i>user busy</i>	486 Busy Here	486
18	<i>no user responding</i>	480 Temporarily Unavailable	408
19	<i>no answer from the user</i>	480 Temporarily Unavailable	480
20	<i>subscriber absent</i>	480 Temporarily Unavailable	480
21	<i>call rejected</i>	480 Temporarily Unavailable	403/603
22	<i>number changed</i>	410 Gone	301
23	<i>redirection to new destination</i>	–	–
25	<i>exchange routing error</i>	480 Temporarily Unavailable	410
27	<i>destination out of order</i>	502 Bad Gateway	502
28	<i>invalid number format (address incomplete)</i>	484 Address Incomplete	484
29	<i>facility rejected</i>	500 Server Internal Error	501
31	<i>normal unspecified</i>	480 Temporarily Unavailable	480

Table 6.4: Q.850 Cause Value to SIP Response Mapping - Normal Class

Q.850	Description	Q.1912.5 SIP Response	RFC3398
34	<i>resource unavailable</i>	480 Temporarily Unavailable	503
34	<i>resource unavailable</i>	486 Busy Here (CCBS possible)	503
38-47	<i>resource unavailable</i>	500 Server Internal Error	503
50	<i>requested facility not subscribed</i>	500 Server Internal Error	–
55	<i>incoming calls barred within CUG</i>	500 Server Internal Error	403
57	<i>bearer capability not authorized</i>	500 Server Internal Error	403
58	<i>bearer capability not presently available</i>	500 Server Internal Error	503
63	<i>service or option not available, unspecified</i>	500 Server Internal Error	–

Table 6.5: Q.850 Cause Value to SIP Response Mapping - Resource Unavailable Class

Q.850	Description	Q.1912.5 SIP Response	RFC3398
65-79	<i>service or option not implemented</i>	500 Server Internal Error	
65	<i>bearer capability not implemented</i>	500 Server Internal Error	488
70	<i>only restricted digital information available</i>	500 Server Internal Error	488
79	<i>service or option not implemented</i>	500 Server Internal Error	501
87	<i>user not member of CUG</i>	500 Server Internal Error	403
88	<i>incompatible destination</i>	500 Server Internal Error	503
90	<i>non-existent CUG</i>	500 Server Internal Error	–
91	<i>invalid transit network selection</i>	404 Not Found	–
95	<i>invalid message, unspecified</i>	500 Server Internal Error	–
97	<i>message type non-existent or not implemented</i>	500 Server Internal Error	–
99	<i>information element/parameter non-existent or not implemented</i>	500 Server Internal Error	–
102	<i>recovery on timer expiry</i>	480 Temporarily Unavailable	504
103	<i>parameter non-existent or not implemented, passed on</i>	500 Server Internal Error	–
110	<i>message with unrecognized parameter, discarded</i>	500 Server Internal Error	–
111	<i>protocol error, unspecified</i>	500 Server Internal Error	500
127	<i>interworking unspecified</i>	480 Temporarily Unavailable	500

Table 6.6: Q.850 Cause Value to SIP Response Mapping - Service or Option Not Available Class

```

SIP/2.0 500 Server Internal Error
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.30
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
      ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=34 ;text="resource unavailable"
Reason: SIP ;cause=500 ;text="server internal error"
Content-Length: 0
Message 6.45: Q.850 Cause Value to SIP Response Mapping Example

```

ANSI Cause Values

The mapping of ANSI cause values returned in an ACM or REL message to SIP responses is illustrated: for the normal class of cause values, in [Table 6.7](#); resource unavailable class, [Table 6.8](#); service or

option not available class, [Table 6.9](#).⁹ These ANSI cause values will be mapped to the corresponding SIP response listed in the tables. The SIP 4xx or 5xx response, CANCEL or BYE message will include the Reason header that contains the ANSI cause value, as well as a Q.850 Reason header that contains the corresponding Q.850 cause value. An example is provided in [Message 6.46](#).

ANSI	Description	SIP Response
23	<i>unallocated destination number</i>	404 Not Found
24	<i>unknown business group</i>	404 Not Found
25	<i>exchange routing error</i>	480 Temporarily Unavailable
26	<i>misrouted call to a ported number</i>	500 Server Internal Error
27	<i>number portability Query on Release (QoR) number not found</i>	500 Server Internal Error

Table 6.7: ANSI Cause Value to SIP Response Mapping - Normal Class

ANSI	Description	SIP Response
45	<i>preemption</i>	500 Server Internal Error
46	<i>precedence call blocked</i>	500 Server Internal Error

Table 6.8: ANSI Cause Value to SIP Response Mapping - Resource Unavailable Class

ANSI	Description	SIP Response
51	<i>call type incompatible with service request</i>	500 Server Internal Error
54	<i>call blocked due to group restrictions</i>	500 Server Internal Error

Table 6.9: ANSI Cause Value to SIP Response Mapping - Service or Option Not Available Class

```
SIP/2.0 500 Server Internal Error
Via: SIP/2.0/UDP cp.heramax.com:5060;branch=z9hG4bK-000001;received=192.168.9.30
Via: SIP/2.0/UDP ss1.customer1.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/UDP client.customer1.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.customer1.com;user=phone>
;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.customer1.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@customer1.com
CSeq: 1 INVITE
Reason: Q.850 ;cause=1 ;text="unallocated (unassigned) number"
Reason: SIP ;cause=404 ;text="not found"
Content-Length: 0
Message 6.46: ANSI Cause Value to SIP Response Mapping Example
```

6.3.4.2 Examples of Unsuccessful Calls from SIP to PSTN

⁹ Note that this mapping is consistent with *Table 21/Q.1912.5*.

6.4 ENUM Calls from PSTN to SIP

A viable option to using the *C-SCP* platform for calls terminating on the SIP network from the PSTN is to use an I-ENUM or P-ENUM DNS server hierarchy. In this approach, rather than using the *C-SCP* platform as a redirect server, the *Optranex 248* directly queries an ENUM database (e.g. bind9, powerdns). Either a host in the internal network provides a dedicated DNS server, or the *Optranex 248* platform acts as a primary DNS server and delegates to ENUM servers maintained by (or on behalf of) the Customer.

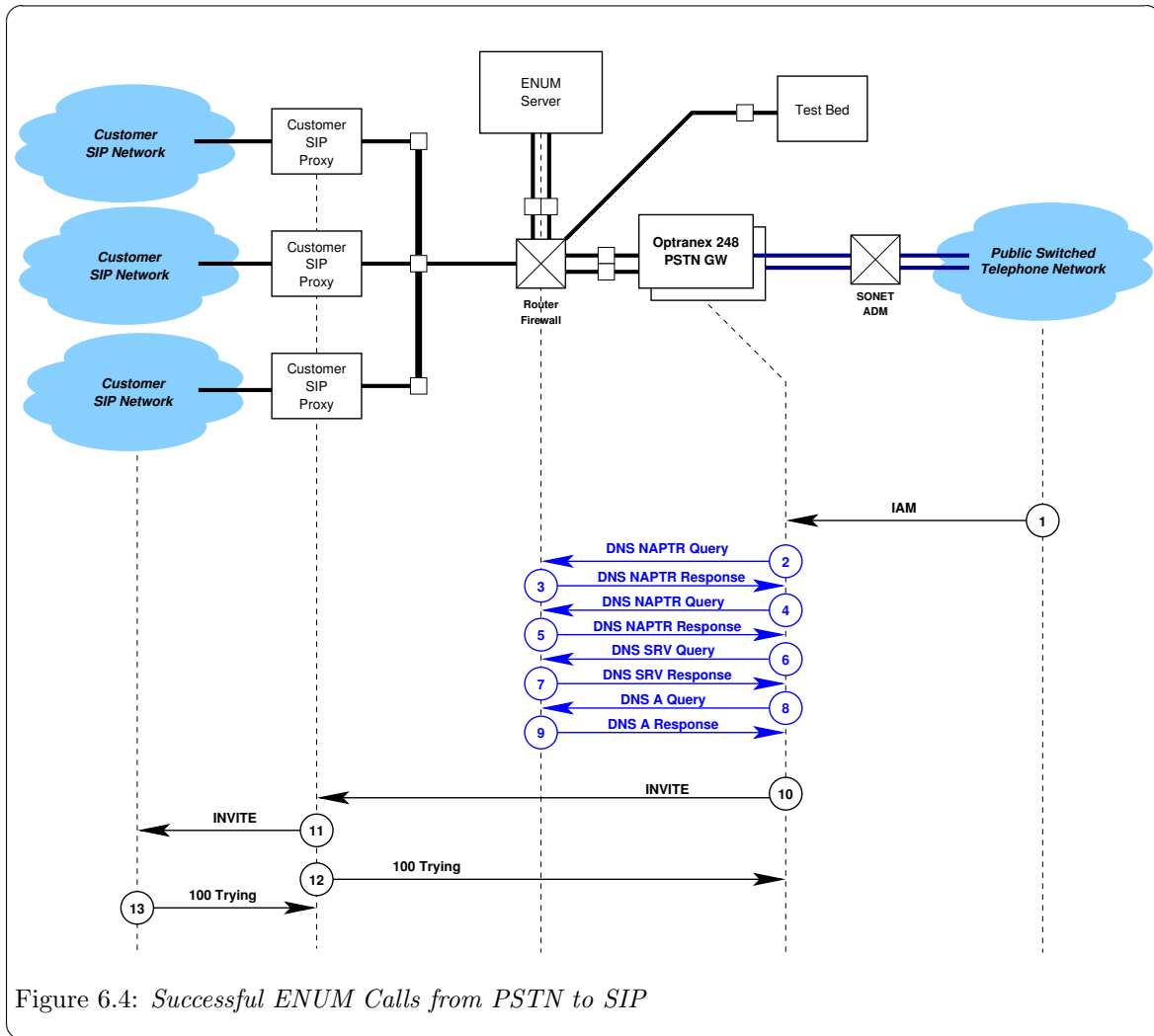
Advantages of this arrangement over the 305 Use Proxy or 303 Proxy Redirect approach with the *C-SCP* platform are as follows:

1. ENUM provides distributed management of mapping from terminating number to **sip** URI through delegation. Customer records can be delegated to the customer.
2. ENUM provides for local caching of transformations at the *Optranex 248*. Local caching of results using the 305 Use Proxy or 303 Proxy Redirect is not possible. A 302 Moved Temporarily message instead of a 305 Use Proxy or 303 Proxy Redirect could be used to provide caching using the **Expires** header or the *expires* parameter in the **Contact** headers in the 302 Moved Temporarily response, however, this mechanism provides no real help with the management of caching as does the ENUM DNS-based approach.
3. ENUM provides for IXFR (incremental zone transfers) which would allow the *Optranex 248* to slave off of the external distributed (even customer managed) ENUM servers and permit per-call lookups to be local to the *Optranex 248*.
4. If the DNS SRV approach is used to distribute calls over like customer proxies for redundancy, a DNS query will be required per call anyway. See *draft-ietf-speermint-srv-naptr-use-06*.
5. The *Optranex 248* does not appear to need to know any of the information on the *C-SCP* platform for the purpose of terminating calls: it just needs to know the **sip** URI associated with the B Number and the results of a lookup attempt.

6.4.1 ENUM Call Flows from PSTN to SIP

6.4.1.1 Successful ENUM Call Flow from PSTN to SIP

A successful ENUM call flow from the PSTN to SIP is illustrated in [Figure 6.4](#).



- Point (1) in [Figure 6.4](#).

A call terminating from the PSTN to the *Optranex 248* switch begins with the receipt of an ANSI ISUP IAM message. For example, see [Example 6.7](#).

<i>FCI</i>	number translated
<i>CgPN</i>	NSN, 222-333-4444, presentation-restricted
<i>CdPN</i>	NSN, 777-666-5555
<i>JIP</i>	830-660
<i>GAP</i>	ported number, NSN, 999-666-5555
<i>GAP</i>	dialed number, NSN, 444-444-4444, presentation-allowed
<i>GAP</i>	additional user provided number - not verified, NSN, 666-666-6666, presentation-allowed
<i>GN</i>	calling name, name available/unknown, "Joe", presentation-allowed
<i>GN</i>	original called name, name available/unknown, "Sue", presentation-allowed
<i>CPC</i>	ordinary subscriber
<i>OLI</i>	ANI 0

Example 6.7: *IAM received for Successful ENUM PSTN to SIP Call*

- Point (2) in [Figure 6.4](#).

The *Optranex 248* uses the *Called Party Number* and potentially the **ported number** in a *Generic Address Parameter* to initiate an ENUM lookup. In the worst case scenario, a DNS NAPTR query is sent to an root ENUM server. The query is, for example:

```
5.5.5.5.6.6.6.9.9.9.e164.arpa
```

Note that this query and response are not required if either the *Optranex 248* has a cached result for the query, or accepts zone transfers from the master or customer ENUM database.

- Point (3) in [Figure 6.4](#).

The root ENUM server performs a lookup which either results in a **sip** URI or an error response. If an error response is returned, the *Optranex 248* can immediately release the call using an ANSI ISUP REL message with the appropriate cause value. When a successful response is received, the *Optranex 248* selects those NAPTR RRs that support E2U resolution service. For example:

```
$ORIGIN 5.5.5.5.6.6.6.9.9.9.e164.arpa.
NAPTR 10 100 "u" "E2U+pstn:tel"
"!^.*$!sip:sue@customer1.com!".
```

Note that this query and response are not required if either the *Optranex 248* has a cached result for the query, or accepts zone transfers from the master or customer ENUM database.

- Point (4) in [Figure 6.4](#).

If a **sip** URI is returned with a domain name portion that does not correspond to an A pointer; a DNS NAPTR lookup is performed to determine the precise host for this query in accordance with *RFC 2915*. This permits the DNS NAPTR and SRV mechanism to be used to distribute calls over redundant incoming customer proxies in accordance with *RFC 2782*. In this example, the target of the DNS NAPTR query is:

```
customer1.com
```

Note that this query and response are not required if either the *Optranex 248* has a cached result for the query, accepts zone transfers from the master or customer DNS database, or the **sip** URI contains a host portion that resolves to an A pointer or a known host to IP address mapping at the *Optranex 248* (e.g. */etc/hosts* entry).

- Point (5) in [Figure 6.4](#).

In response to the DNS NAPTR query, the root DNS server responds with series of "s" substitutions. In this example, the zone records might look like:

```
IN NAPTR 50 50 "s" "SIPS+D2T" "" _sips._tcp.customer1.com
IN NAPTR 90 50 "s" "SIP+D2T" "" _sip._tcp.customer1.com
IN NAPTR 100 50 "s" "SIP+D2U" "" _sip._udp.customer1.com
```

Note that this query and response are not required if either the *Optranex 248* has a cached result for the query, accepts zone transfers from the master or customer DNS database, or the **sip** URI contains a host portion that resolves to an A pointer or a known host to IP address mapping at the *Optranex 248* (e.g. */etc/hosts* entry).

- Point (6) in [Figure 6.4](#).

The *Optranex 248* selects a target from the response to the NAPTR DNS query on the target domain and generates a DNS SRV query on the service target. In this example, the *Optranex 248* only supports UDP and therefore selects the entry:

```
_sip._udp.customer1.com
```

If the previous DNS NAPTR lookup results in no records, a normal DNS SRV lookup for ‘_sip._udp.customer1.com’ is performed instead.

Note that the caching period on NAPTR SRV entries is typically 1 to 12 hours permitting for infrequent refresh of the information. Should the customer DNS (or authoritative DNS) be configured for zone transfers to the *Optranex 248*, the interval can be extended indefinitely.

- Point (7) in [Figure 6.4](#).

The response to the DNS SRV contains either a DNS A record or AAAA record, or does not (e.g. it contains a CNAME record). If the response contains an A or AAAA record, the IP address of the record is used directly. Otherwise, *RFC 2782* is followed and an A record results. In this example assume that a CNAME record results as follows:

```
$ORIGIN customer1.com
_sip._udp SRV 0 1 5060 proxy1.customer1.com
_sip._udp SRV 0 1 5060 proxy2.customer1.com
```

Note that the caching period on SRV entries is typically on the order of days permitting for very infrequent refresh of the information. Should the customer DNS (or authoritative DNS) be configured for zone transfers to the *Optranex 248*, the interval can be extended indefinitely.

- Point (8) in [Figure 6.4](#).

When there is no A nor AAAA record in the response to the DNS SRV query, the procedures of *RFC 2782* are followed to select a record look up an A or AAAA record from, for example, a CNAME record. In this example, the target of the DNS query is the CNAME from the SRV response:

```
proxy1.customer1.com
```

Note that if no records were returned to the previous DNS SRV request, a normal A request would be made on just domain target ‘customer1.com’.

Note that caching period on A records can be days causing very infrequent lookups of the information.

- Point (9) in [Figure 6.4](#).

The response on the query to the specific customer proxy will result (when successful) an an A or AAAA record which can then be used as the IP address of the proxy.

```
$ORIGIN customer1.com
proxy1 A 172.30.79.10
proxy2 A 172.30.79.11
```

- Point (10) in [Figure 6.4](#).

The *Optranex 248*, using the results from the *NAPTR* and *SRV* lookups (whether cached, slaved or remote), formulates an *INVITE* to the **sip** URI resulting from the lookup and directs it at the customer's incoming proxy (indicated in the **sip** URI or DNS *SRV* result from the lookup).

```
INVITE sip:sue@customer1.com SIP/2.0
Via: SIP/2.0/UDP gw1.heramax.com:5060;branch=z9hG4bK-000001
From: "Joe" <tel:+16666666666>;tag=XXX
To: "Sue" <tel:+14444444444>
Call-ID: 111223@gw1.heramax.com
CSeq: 1 INVITE
Contact: <sip:+1666666666@gw1.heramax.com:5060;user=phone;transport=udp>
Max-Forwards: 33
Content-Type: application/sdp
Content-Length: 134

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.heramax.com
s=-
c=IN IP4 gw1.heramax.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Message 6.47: ENUM Customer INVITE for Successful PSTN to SIP Call
```

- Point (11) in [Figure 6.4](#).

The customer proxy determines the location of the destination UA and forwards the *INVITE* using *RFC 3261* procedures.

- Point (12) in [Figure 6.4](#).

A *100 Trying* is potentially returned and the call completes or is treated according to *RFC 3261* procedures.

- Point (13) in [Figure 6.4](#).

A *100 Trying* is potentially returned from the UA and the call completes or is treated according to *RFC 3261* procedures.

The ENUM call flow has the significant advantage that, with proper ENUM and DNS management, the majority of calls will not have to query outside the *Optranex 248*. With 64,512 circuits at an average 3 minute holding time, the resulting 360 calls per second can avoid 4 message exchanges with the network per call. Also, terminating calls do not rely upon the availability of an external box. If the root ENUM/DNS server should go down, cached and slaved results in the ENUM/DNS server local to the *Optranex 248* can be used. As incremental zone transfers might stop during a root or customer ENUM/DNS server outage, some calls terminating from the PSTN might be directed to an incorrect customer proxy; however, the customer proxy will be able to treat those calls and return appropriate error codes to the *Optranex 248* for releasing the call with the proper cause values back to the PSTN.

Example BIND9 Zone Files

An example of a `bind9` configuration file is illustrated in [Example 6.8](#).

```
acl good_sources { 192.168.0.0/24; };

options {
    ...
    also-notify      { 192.168.0.10; 192.168.0.11; };
    allow-query      { localhost; good_sources; };
    allow-transfer    { localhost; good_sources; };
    allow-recursion  { localhost; good_sources; };
    ...
};

...

zone "1.e164.heramax.com" IN {
    type master;
    file "/etc/bind/db.1.e164.heramax.com";
    allow-query { any; };
    allow-transfer { good_sources; };
    notify yes;
    also-notify { 192.168.0.10; 192.168.0.11; };
    zone-statistics yes;
};
```

Example 6.8: *Example BIND9 configuration file (named.conf.local)*

An example of a `bind9` zone file is illustrated in [Example 6.9](#).

```

$TTL 86400
$ORIGIN heramax.com.
@ IN SOA ns.heramax.com. root.ns.heramax.com. (
    2009013000
    28800
    14400
    3600000
    86400 )
NS ns.heramax.com.
IN NAPTR 50 50 "s" "SIPS+D2T" "" _sips._tcp.heramax.com.
IN NAPTR 90 50 "s" "SIP+D2T" "" _sip._tcp.heramax.com.
IN NAPTR 90 50 "s" "SIP+D2U" "" _sip._udp.heramax.com.
_sips._tcp SRV 0 1 5060 gw1.heramax.com.
_sips._tcp SRV 0 1 5060 gw2.heramax.com.
_sip._tcp SRV 0 1 5060 gw1.heramax.com.
_sip._tcp SRV 0 1 5060 gw2.heramax.com.
_sip._udp SRV 0 1 5060 gw1.heramax.com.
_sip._udp SRV 0 1 5060 gw2.heramax.com.
gw1 IN A 192.168.0.10.
gw2 IN A 192.168.0.11.

```

Example 6.9: *Example BIND9 zone file*

An example of a bind9 zone file is illustrated in [Example 6.10](#).

```

$ORIGIN 1.e164.heramax.com.
@ IN SOA ns.heramax.com. root.ns.heramax.com. (
    2009013000
    28800
    14400
    3600000
    86400 )
NS ns.heramax.com.
1.1.0.9.4.0.8.7 3600 IN NAPTR 100 10 "u" \
    "E2U+sip" "!"^(.*)$!sip:\\1@openss7.net!" .
1.1.0.9.4.0.8.7 3600 IN NAPTR 100 10 "u" \
    "E2U+pstn:tel" "!"^(.*)$!tel:\\1;npdi;phone-context=openss7!" .

```

Example 6.10: *Example BIND9 zone file*

6.4.1.2 Unsuccessful ENUM Call Flow from PSTN to SIP

When the ENUM call flow from PSTN to SIP is unsuccessful, it can be treated in four locations:

1. At the *Optranex 248* as a result of a local NAPTR or SRV lookup that is cached or slaved. When an error result is returned to either of the DNS lookups, the error result can be mapped onto ANSI ISUP *Cause* values in the REL message.
In contrast to the *C-SCP* redirect server approach, where it is not possible for the *Optranex 248* to treat the call immediately from local data, the ENUM/DNS approach permits this. Although calls that go to treatment by far the minority of calls, this saves additional delay and additional trunk holding times after seizure.
2. At the ENUM/DNS root server as a result of a local NAPTR or SRV lookup. When an error

result is returned to either of the

3. At the Customer Proxy as a result of proxy operation.
4. At the Customer UA.

6.5 ENUM Calls from SIP to PSTN

A viable option to using the *C-SCP* platform for analyzing and routing calls terminating on the PSTN from the SIP network is to use the same I-ENUM or P-ENUM DNS server hierarchy presented in [Section 6.4 \[ENUM Calls from PSTN to SIP\], page 83](#). In this approach, rather than using the *C-SCP* platform as a redirect server, the *Optranex 248* directly queries an ENUM database (e.g. bind9, powerdns). Either a host in the internal network provides a dedicated DNS server, or the *Optranex 248* platform acts as a primary DNS server and delegates to ENUM servers maintained by (or on behalf of) the Customer.

Advantages of this arrangement over the **302 Moved Temporarily** approach with the *C-SCP* platform are as follows:

1. ENUM provides distributed management of mapping from terminating number to **sip** URI through delegation. Customer records can be delegated to the customer.
2. ENUM provides for local caching of transformations at the *Optranex 248*. Local caching of results using the **302 Moved Temporarily Expires** header or the *expires* parameter in the **Contact** headers in the **302 Moved Temporarily** response, however, this mechanism provides no real help with the management of caching as does the ENUM DNS-based approach.
3. ENUM provides for IXFR (incremental zone transfers) which would allow the *Optranex 248* to slave off of the external distributed (even customer managed) ENUM servers and permit per-call lookups to be local to the *Optranex 248*.

6.5.1 ENUM Call Flow from SIP to PSTN

1. The SIP phone within the customer network generates an **INVITE** message toward the customer proxy.
2. The customer proxy authenticates the SIP phone user and determines that the call is to be routed via the *Optranex 248* gateway switch and forwards the **INVITE**.
3. The *Optranex 248* gateway switch receives the **INVITE** and initiates call processing consisting of the following steps:
 1. The format of the **From**, **P-Requested-Identity**, **P-Asserted-Identity**, **To** and **Request-URI** headers are examined for proper format.
 2. After determining a proper E.164 number associated with the originator, the number is used to perform an ENUM lookup. The gateway examines the response for '**E2U+pstn:tel**' and '**E2U+SIP**' records. The '**E2U+SIP**' response record is examined to ensure that the domain portion of the **sip** URI contains an inbound customer proxy that corresponds to the outbound customer proxy from which the **INVITE** message was received (authorization and verification). The *phone-context* parameter in the '**E2U+pstn:tel**' record is used to establish the originating class of service (MTA), and any *rn* in the **tel** URI can be used to establish the *JIP*. The *cpc* and *oli* can be used to provide charging information. The **tel** URI may also contain a **P-Charge-Info** header substitutions which can contain a billing (charge) number. (When the *oli* is included without a header substitution, the calling number is used by default by ANSI ISUP as the charge number.)
 3. After determining a proper E.164 number associated with the termination, the number is used to perform an ENUM number portability lookup. The resulting '**E2U+pstn:tel**' record contains the number portability dip indicator and the routing number (if applicable).
 4. After routing the destination number (whether called number or local routing number), the gateway performs an I-ENUM lookup on the destination number and obtains a

- ‘E2U+pstn:tel’ record containing the terminating MTA and OCN in the *phone-context* parameter. This value is used to establish the terminating class of service (MTA, OCN).
5. Having established the originating class of service and the terminating class of service, a P-ENUM lookup is performed on a screening ENUM tree for the destination number. The ‘E2U+pstn:tel’ record in the response can contain a *tsn* (Transit Network Selector) parameter or a *trgp* and *trunk-context* parameter. These parameters are used to establish the outgoing trunk group associated with the transit network.
 4. Finally, the *Optranex 248* gateway switch formulates an ANSI ISUP IAM and launches it toward the selected transit network on the selected trunk group. The rest of the call completes as normal.

This arrangement provides for the option of placing a voice-peering proxy between the customer proxy and the gateway. This proxy can perform its own I-ENUM lookups for the voice peering exchange and route SIP calls over the voice peering fabric, only passing calls to the gateway when the voice peering fabric is not available. Also, when multiple gateways are involved, the proxy can select the appropriate gateway using any number of factors. When this proxy performs a number portability dip, it can include the *npdi* and possibly *rn* in the Request-URI so that the gateway does not have to redip for NP. Also, if a P-ENUM dip is also performed (for access restrictions and verification on the originator, or for the termination), a *P-Asserted-Identity* field and the *Request-URI* forwarded to the gateway can contain the associated parameters and the *enumdi* parameter indicating that another P-ENUM dip is not required.

As a fall-back for the NP ENUM dip, the gateway can launch an SS7 LNP dip or perform the 302 Moved Temporarily procedure with the *C-SCP* when necessary.

6.5.1.1 ENUM Tables (DNS Zones)

The following ENUM trees are required by the *Optranex 248* gateway to implement the call flow as described above:

- P-ENUM base tree providing ‘E2U+SIP’ entries for calls terminating to customer SIP networks from the PSTN (used also in the reverse direction for verification and authorization), and ‘E2U+pstn:tel’ entries for establishing the *phone-context*, *rn*, *oli*, *cpc* and P-Charge-Info header substitutions for call origination through the gateway.
- NP ENUM tree providing ‘E2U+pstn:tel’ entries for number portability with *npdi* and *rn* parameters as required.
- P-ENUM tree partitioned by originating class of service (MTA) providing ‘E2U+pstn:tel’ entries providing *phone-context*, *tns*, *trgp* and *trunk-context* parameters for routing.

P-ENUM Tree

```
*.e164.heramax.com (base ENUM tree)
*.[mta2].[mta1].e164.heramax.com (screening ENUM tree)
```

NP ENUM Tree

```
*.e164.arpa (user ENUM tree)
```


7 Software Architecture

The software architecture of the *Optranex 248* switch contains select components of the *OpenSS7* STREAMS and Signaling Protocol Stacks. Also included are the *Optranex Incorporated* firmware and software drivers for the *Optranex 248*.

7.1 General Software Architecture

The software architecture of the *Optranex 248* switch uses primarily the protocol and media processing components of the *OpenSS7 Project*. Additional components include the *sofia-SIP* SIP stack and the *OpenNMS net-snmp* SNMP agent framework. The software architecture is decomposed into the following major components:

1. The *Signaling Stack* provides the ISUP/SIGTRAN signaling functions for use by the *Optranex 248* switch (see [Section 7.1.1 \[Signaling Stack\]](#), page 94).
2. The *Media Stack* provides the SONET/SDH and RTP media handling functions as well as the integrated media gateway functions of the *Optranex 248* switch (see [Section 7.1.2 \[Media Stack\]](#), page 96).
3. The *Application Subsystem* provides the *sofia-SIP* SIP stack and SIP-ISUP interworking functionality (see [Section 7.1.3 \[Application Subsystem\]](#), page 98).
4. The *Management Subsystem* provides the active *net-snmp* SNMP agents required to configure, measure, monitor, alarm and manage the signaling and media stacks in the *Optranex 248* (see [Section 7.1.4 \[Management Subsystem\]](#), page 100).

7.1.1 Signaling Stack

An overview of the signaling stack is illustrated in [Figure 7.1](#).

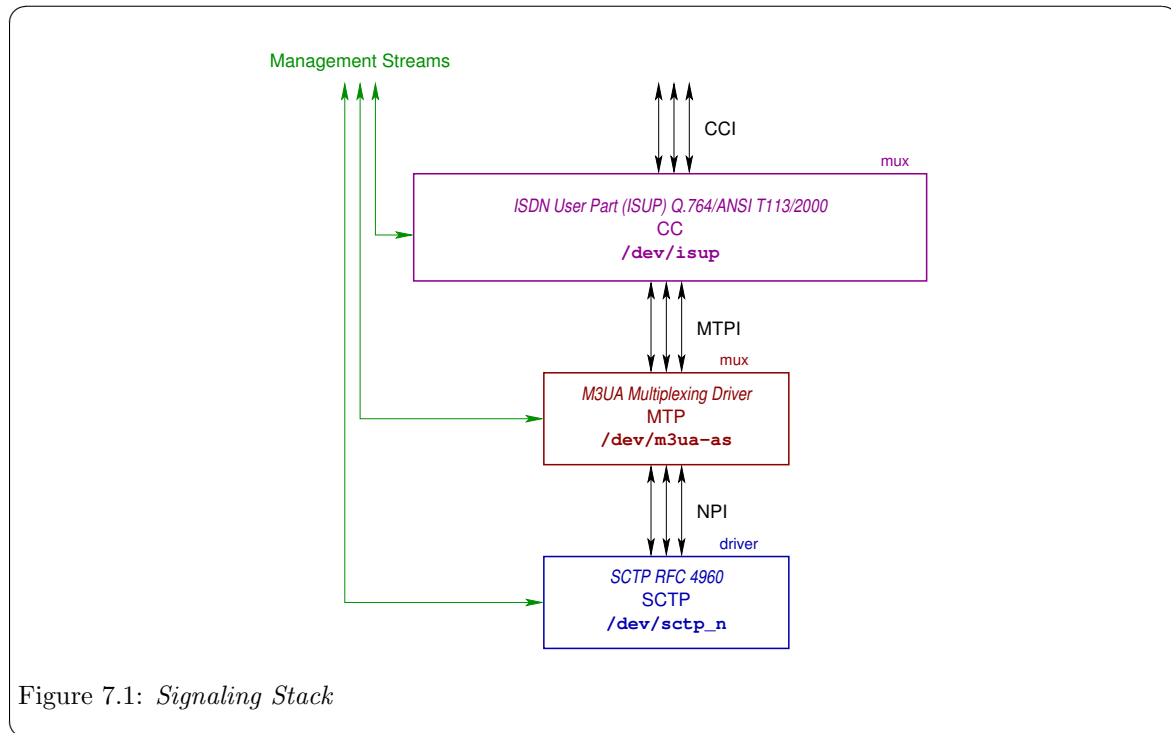


Figure 7.1: *Signaling Stack*

The signaling stack consists of the following primary subsystems:

1. An ISUP driver (see [Section 7.1.1.1 \[ISUP Driver\]](#), page 94).
2. The SIGTRAN stack providing SS7 connectivity for the ISUP driver (see [Section 7.1.1.2 \[SIGTRAN Stack\]](#), page 94).

7.1.1.1 ISUP Driver

As illustrated in [Figure 7.1](#), the *ISUP Driver* consists of a monolithic pseudo-device multiplexing driver, `/dev/isup`, that implements the *ISDN User Part* of the SS7 stack in accordance with ITU-T Recommendation Q.764 and ANSI T1.113/2000 as well as several other variants. The driver is responsible for performing ISUP signaling for the *Optranex 248* switch. The upper service interface is the *Call Control Interface* (CCI) developed by *OpenSS7* and documented in the CCI specification. The lower service interface is the *Message Transfer Part Interface* (MTPI) which is capable of connecting both to *OpenSS7*'s traditional SS7 stack implementation of MTP as well as *OpenSS7*'s SIGTRAN M3UA stack, in a transparent fashion.

The *ISUP Driver* provides management streams for use by the active SNMP agent that is responsible for configuring, measuring, monitoring and maintaining ISUP signaling.

7.1.1.2 SIGTRAN Stack

As illustrated in [Figure 7.1](#), the *SIGTRAN Stack* consists of a sequence of pseudo-device and multiplexing drivers that provide all of the SIGTRAN connectivity for the signaling stack.

M3UA Driver

The *M3UA Driver* is the *OpenSS7* STREAMS implementation of the AS-side of the M3UA protocol as specified in RFC 4666. Two AS are implemented, one for communications with an SG in the SS7 network; another for A-side/B-side communications between ISUP drivers on the duplicated platforms. The driver provides a *Message Transfer Part Interface* (MTPI) at its upper service interface which is *OpenSS7*'s implementation of the Q.704/ANSI T1.111.4 service primitive interface. The driver provides a *Network Provider Interface* (NTPI) at its lower service interface for communications with the underlying *SCTP Driver*.

The driver provides management streams for use by the active SIGTRAN SNMP agent that is responsible for configuration, measurements, monitoring, auditing and controlling M3UA using the *OpenSS7* enterprise MIB for M3UA.

SCTP Driver

The *SCTP Driver* is the *OpenSS7* STREAMS implementation of SCTP. This is a high-performance concurrent multipath transfer driver that implements RFC 4960. The driver provides either a *Network Provider Interface* (NPI) or *Transport Provider Interface* (TPI) at its upper service interface.

Configuration and management of the driver is performed by the active M3UA SNMP agent.

7.1.2 Media Stack

An overview of the media stack is illustrated in [Figure 7.2](#).

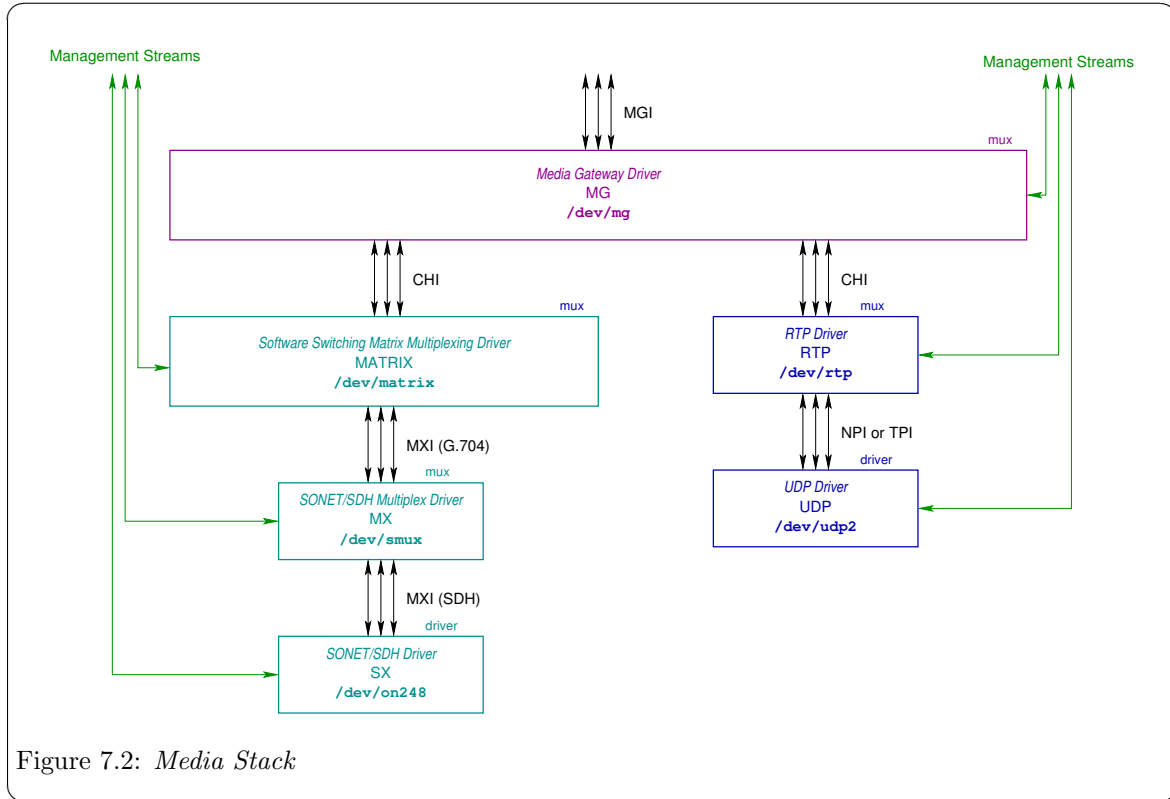


Figure 7.2: *Media Stack*

The media stack consists of the following primary subsystems:

1. A *Media Gateway Driver* that is responsible for the overall media gateway functions (see [Section 7.1.2.1 \[Media Gateway Driver\]](#), page 96).
2. A *TDM Stack* consisting of a sequence of components that provide for the TDM portions of the media gateway functions (see [Section 7.1.2.2 \[TDM Stack\]](#), page 97).
3. A *Ephemeral Stack* consisting of a sequence of components that provide for the Ephemeral portions of the media gateway functions (see [Section 7.1.2.3 \[Ephemeral Stack\]](#), page 98).

These primary subsystems are described in the sections that follow. All subsystems, with the exception of the *Optranex 248 SONET/SDH Driver* are licensed under the [\[GNU Affero General Public License\]](#), page 173 and the [\[GNU General Public License\]](#), page 183. The *Optranex 248 SONET/SDH Driver* is distributed under a proprietary license and is protected by pending patents.

7.1.2.1 Media Gateway Driver

As illustrated in [Figure 7.2](#), the *Media Gateway Driver*, `/dev/mg`, is responsible for performing all MG functions and marshalling media samples between the *TDM Stack* and the *Ephemeral Stack*. Control of the media gateway is performed using the *Media Gateway Interface* (MGI).¹ The MGI performs a full set of MG functions using a STREAMS based service primitive interface. The MG

¹ The *Media Gateway Interface* is documented in the *OpenSS7 MGI Specification*.

functions provided are equivalent to those provided by *ITU-T Recommendation H.248*. In particular, the MGI is capable of the following:

- forming media connections between TDM and ephemeral media streams
- playing tones and announcements
- generating DTMF digits
- applying loopbacks for continuity testing
- continuity tone
- milliwatt
- silent termination
- balanced termination
- ringback patterns
- busy patterns (T60)
- reorder patterns (T120)
- receiver off hook patterns (ROH)
- bong tone

The *Media Gateway Driver* provides for control of the MG functions using the MGI. The driver also provides management streams for use by a user-space active SNMP agent that is capable of measuring, monitoring, auditing and maintaining the media gateway functions using *OpenSS7* enterprise MIBs.

7.1.2.2 TDM Stack

As illustrated in [Figure 7.2](#), the *TDM Stack* consists of a sequence of device and pseudo-device multiplexing drivers that provide all of the TDM connectivity for the media gateway.

Software Switching Matrix Multiplexing Driver

The *Software Switching Matrix Multiplexing Driver*, `/dev/matrix`, is responsible for performing channel switching between the TDM interface components and the *Media Gateway Driver*. The major purpose of this module is software TDM switching. Not all channels associated with the TDM interfaces are necessarily under the control of the media stack: some channels may be switched to other modules and drivers by the matrix driver. For example, if some channels are used for traditional SS7 connectivity, they would be mapped to a different stream on the upper service interface that would connect into the bottom of the *OpenSS7* traditional SS7 stack.

SONET/SDH Multiplex Driver

The *SONET/SDH Multiplex Driver*, `/dev/smux`, is responsible for extracting TDM spans from one or more SONET/SDH synchronous payload envelopes. This is performed by extracting spans from VT1.5 or VT2 virtual tributaries within the SPE, or by extracting an demultiplexing DS3 or E2 payloads into their constituent T1 or E1 spans (i.e. 3:1 multiplexing). It is also responsible for monitoring segment, path and span carrier conditions. The upper service interface is a *Multiplex Interface* (MXI) referenced to G.704. The lower service interface is also a *Multiplex Interface* (MXI); however, the lower service interface is referenced to SONET/SDH (i.e. synchronous payload envelopes instead of T1 or E1 frames).

This driver also provides for management streams that are used to measure, monitor, audit and maintain the SONET/SDH segment and path, as well the constituent

DS3, E2, T1 and E1 facilities. A user-space active SNMP agent is responsible for configuration and management using this stream.

SONET/SDH Driver

The *SONET/SDH Driver*, `/dev/on248`, is the device driver for the *Optranex 248* SONET/SDH interface card. This driver is primarily responsible for marshalling SONET/SDH synchronous payload envelopes to and from the optics. Note that not all SPE are necessarily mapped into the `/dev/smux` driver: some SPE (possibly concatenated STS or VC) may be mapped to other upper service interfaces that are used for PPP over SONET or GFP for passing IP or Ethernet to or from the synchronous network.

This driver also provides for management streams that are used to measure, monitor, audit and maintain the SONET/SDH segment and path, as well as the performance of hardware devices (such as measurements and management of the optical interface modules, lasers and automatic protection switching).

7.1.2.3 Ephemeral Stack

As illustrated in [Figure 7.2](#), the *Ephemeral Stack* consists of a sequence of pseudo-device drivers and multiplexing drivers that provide all of the ephemeral (RTP) connectivity for the media gateway.

RTP Driver

The *RTP Driver* is responsible for marshalling media samples to and from the `/dev/mg` driver using the *Channel Interface* (CHI) provided on the upper service interface. These samples are inserted into or extracted from RTP packets in accordance with the codecs specified to it by the `/dev/mg` driver. This driver also performs all necessary jitter buffering and monitors for loss of RTP connectivity.

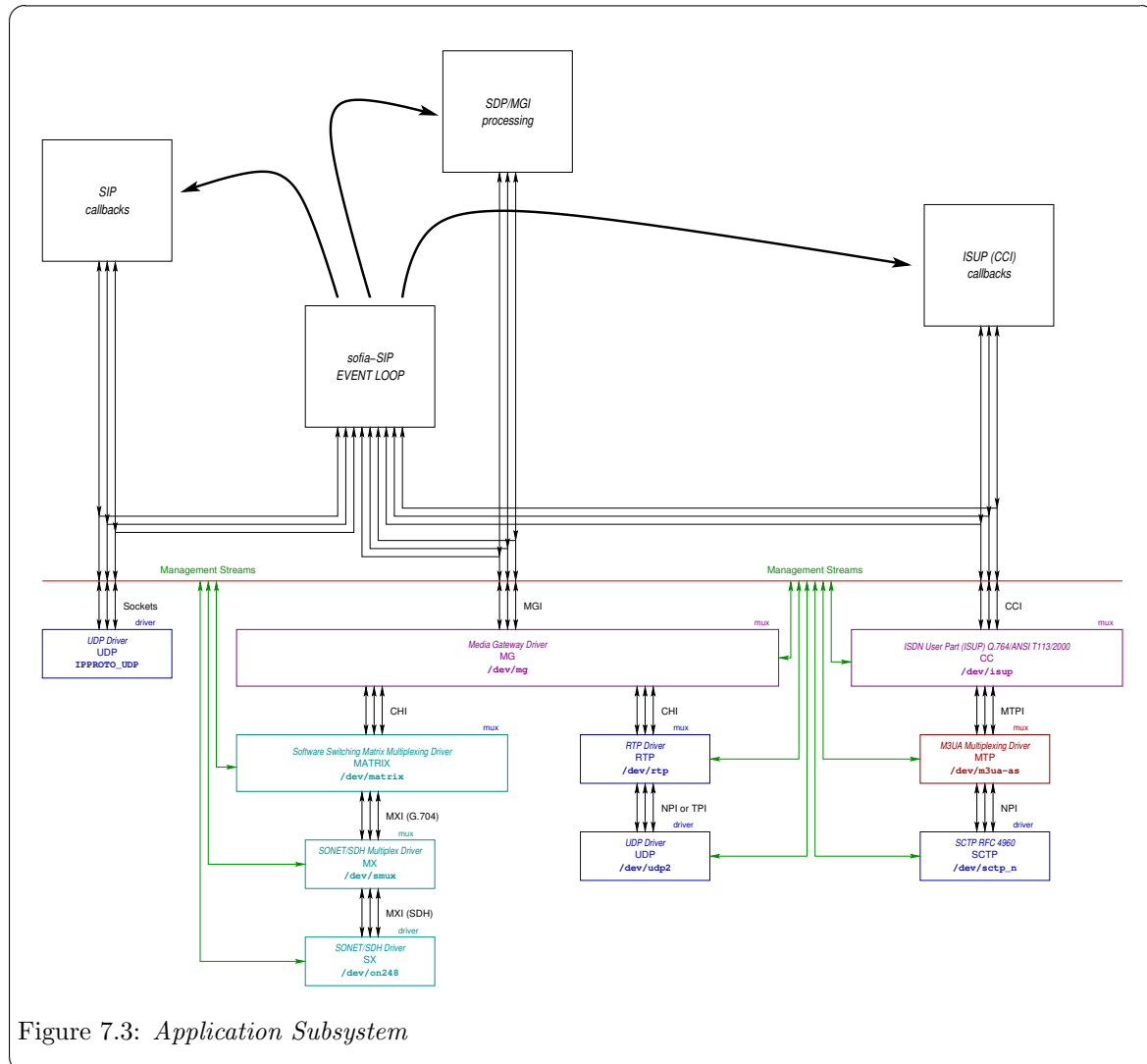
The driver also provides for management streams that are used to measure, monitor, audit and maintain the RTP connections.

UDP Driver

The *UDP Driver* is responsible for passing RTP packets to and from the network. This component uses the *OpenSS7* high performance UDP driver that interfaces directly with the *Linux* IP subsystem.

7.1.3 Application Subsystem

The relationship between the application and the signaling and media stacks is illustrated in [Figure 7.3](#).



The *Application Subsystem* consists primarily of a *Linux* application process that utilizes the *sofia-SIP* framework for performing SIP signaling; the CCI interface provided by the *OpenSS7 Signaling Stack* for performing ISUP signaling; and the MGI interface provided by the *OpenSS7 Media Stack* for media gateway control.

The interworking between SIP and ISUP that is performed by the application subsystem consisting of the CCI interface to the ISUP stack, MGI interface to the media gateway functions, and interface for SIP signaling is detailed in [Section 7.2 \[SIP to ISUP Interworking\]](#), page 101.

Separate application programs and processes will be provided for circuit maintenance and test calls for ISUP. These applications will open streams on the ISUP driver that are dedicated to use for test and maintenance.

The test call program will permit test calls from the PSTN to terminate on the *Optranex 248*. The test call program receives all CCI interface connection indications (corresponding to **IAM** messages) that initiate test calls. The called party number will be examined to determine the test line function that is being invoked and utilize the MGI interface to perform the necessary actions to implement the test call.

The circuit maintenance program will perform continuity test functions that are a normal part of regular calls, as well as continuity test functions that are invoked by remote maintenance. The CCI interface is used for delivery of maintenance primitives and the MGI interface is used by the program to effect continuity test functions.

7.1.4 Management Subsystem

The relationship between management and the signaling and media stacks is illustrated in [Figure 7.4](#).

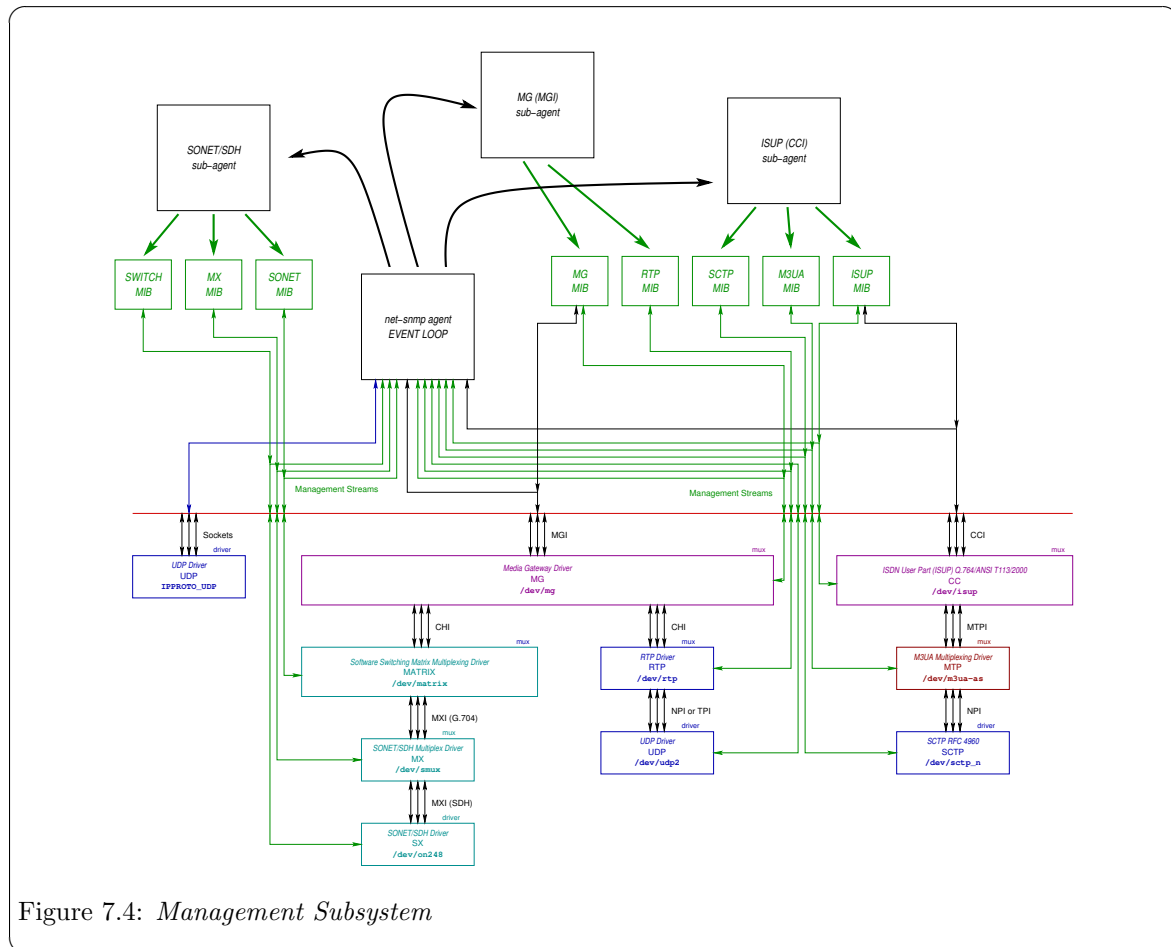


Figure 7.4: *Management Subsystem*

The *Management Subsystem* consists of a number of active SNMP agents implemented using the *net-snmp* framework. The agents utilize management streams from the *OpenSS7 Signaling Stack* and *Media Stack* to perform their functions. The SNMP agents use the self-documenting *OpenSS7* enterprise MIBs.

In addition, the STREAMS framework provides management streams and an active SNMP agent that provides for the management of the STREAMS framework itself using the *OpenSS7* enterprise STREAMS MIB.

For more information on the management architecture, see [Chapter 10 \[Management Architecture\]](#), page 133.

7.2 SIP to ISUP Interworking

This section specifies the interworking of SIP to ISUP signaling. This interworking is internal to the *Optranex 248*. The overall SIP to ISUP interworking is illustrated in [Figure 7.3](#).

The *OpenSS7 ISUP* stack and *Call Control Interface (CCI)* are used for interworking ISUP calls with SIP. The *Optranex 248* uses the *sofia-SIP* stack, developed by *Nokia*, for processing of SIP signaling. The *nta* module is used for interworking concurrent SIP calls with ISUP calls. A purpose-developed interworking function exists at the control nexus between the *sofia-SIP* and *OpenSS7 ISUP* stacks.

OpenSS7 ISUP Stack

The *OpenSS7 ISUP* stack is illustrated in [Figure 7.2](#) and interactions between the interworking function and the *OpenSS7 ISUP* stack are illustrated in [Figure 7.3](#). The *OpenSS7 ISUP* stack is a STREAMS application and API developed by the *OpenSS7 Project* and released under the [\[GNU Affero General Public License\]](#), page 173.

The *OpenSS7 ISUP* stack is a STREAMS application and API that consists primarily of a call control API user that passes primitives to the ISUP stack using formatted ‘C’ language structures which are written to a STREAMS file descriptor using `putmsg(2s)`. Primitives are read from the ISUP stack by reading primitives formatted as ‘C’ language structures which are read from a STREAMS file descriptor using `getmsg(2s)`. The first field of each primitive contains a primitive type which corresponds to a defined primitive for the interface. The remaining members of the ‘C’ language structure are specific to the primitive indicated in the first field. In this way, the interface and API consists of a fully asynchronous message passing mechanism, where the sequence of primitives issued to and received from the interface are constrained only by the interface definition. The interface definition used by the *OpenSS7 ISDN*, *H.323*, *ISUP* and *BICC* stack components is the [Call Control Interface](#).

Nokia sofia-SIP Libraries

The *sofia-SIP* libraries are illustrated in [Figure 7.3](#). The *sofia-SIP* libraries are a set of ‘C’ language libraries developed by *Nokia* and released under the [\[GNU Lesser General Public License Version 2.1\]](#), page 193.¹

The *Nokia sofia-SIP* stack provides a module (subsystem) which is called the *Nokia Transaction Agent* or *nta* for short. The *nta* is a transaction engine, event loop and callback dispatch mechanism for building SIP applications. The libraries provide SIP and SDP message parsing and encoding, TCP and UDP socket event loops, the ability to hook additional file descriptors into the event loop, basic timers and SIP transaction state machine components.

SIP-ISUP Interworking Function

The SIP to ISUP interworking function is illustrated in [Figure 7.3](#). The SIP to ISUP interworking function is written within the scope of a *sofia-SIP nta* application. The interworking function uses the event loop and dispatch mechanisms of the *su* module of the *sofia-SIP* libraries. As with any other event loop based library, some mechanism is provided to permit the callback of a function on typical `select(2)` or `poll(2s)` events.² Because the *OpenSS7 ISUP* stack is based on a STREAMS interface whose file descriptors support the entire range of functionality of `poll(2s)`, including that of STREAMS, it is very easy to simply install the ISUP file descriptor with a `pollfd` structure using the `su_root_register()` function in *sofia-SIP*.

¹ Note that the *sofia-SIP* libraries are released by *OpenSS7* under the [\[GNU General Public License\]](#), page 183 in accordance with [Section 3](#) of the [\[GNU Lesser General Public License Version 2.1\]](#), page 193.

² *net-snmp* only provides `select(2)`.

The interworking function will open three streams (file descriptors resulting from a `open(2s)` call on device `/dev/streams/isup`) for processing call control. One stream will be bound (with the `CC_BIND_REQ` primitive after opening) as a listening streams with the `cc_setup_ind` field of the `CC_BIND_REQ` set to a positive non-zero integer. A second stream will be bound with the `cc_setup_ind` field of the `CC_BIND_REQ` set to zero. A third stream will be bound with the `cc_setup_ind` field of the `CC_BIND_REQ` set to zero. All three streams will be bound to an ISUP address with signaling point scope, `ISUP_SCOPE_SP` using the signaling point code of the *Optranex 248* switch. All incoming call indications will then be received on the first stream. All outgoing call requests will be sent on the second stream. All incoming call indication received on the first stream will be accepted on the third stream. The resulting three file descriptors will each be installed into the *sofia-SIP* event loop using the `su_root_register()` and `su_root_eventmask()` functions. Three separate callback functions will be associated with each stream.

The SIP-ISUP interworking function should have two major modes of operation: one conforming to *RFC 3398* and another conforming to *ITU-T Recommendation Q.1912.5*. These operation modes differ primarily in the cause values and response codes returned under various circumstances. For the Dallas and Seattle trials, *RFC 3398* is preferred initially.

7.2.1 Interworking Calls from SIP to PSTN

All calls from SIP to PSTN are initiated upon the receipt of an INVITE message at the *sofia-SIP* stack. The interworking function establishes an *nta* call leg to establish a transaction. CCI primitives sent to and received from the ISUP stack will be issued and received on the second ISUP stream described above. CCI primitives issued to the ISUP stack will be issued from the *nta* message callbacks. CCI primitives will be received on the installed callback for the second ISUP stream.

7.2.1.1 Mapping SIP INVITE to ISUP IAM

Upon receiving a INVITE message at the *sofia-SIP* stack beginning a transaction, the interworking function establishes a *nta* call leg and transaction, and prepares a `CC_SETUP_REQ` primitive. The interworking function assigns a `cc_user_ref` to identify the call with the ISUP stack. The `cc_call_type` will either be `CC_CALL_TYPE_SPEECH` or `CC_CALL_TYPE_3_1kHz_AUDIO`. The `cc_call_flags` will contain a logical OR of the following flags:

`ISUP_NCI_ONE_SATELLITE_CCT`

`ISUP_NCI_TWO_SATELLITE_CCT`

The `ISUP_NCI_ON_SATELLITE_CCT` flag is set in accordance with Q.1912.5 indicating that there is one satellite in the circuit.

`ISUP_NCI_CONT_CHECK_REQUIRED`

This flag is set or not set depending upon the continuity check requirements of the interconnect. Typically, continuity checks (`ISUP_NCI_CONT_CHECK_REQUIRED`) are performed on one in ‘N’ calls, where ‘N’ is typically ‘10’.

Because the Dallas trial will not initially perform continuity checks on outgoing circuits (continuity checks will be performed, instead, as part of maintenance turn-up of each circuit), this flag will not initially be set. Later, a mechanism will be required for determining on which outgoing calls to set this flag. The determination may be based on an outgoing trunk group basis.

`ISUP_NCI_CONT_CHECK_PREVIOUS`

As the previous circuit is not an ISUP circuit (the call is from the SIP network), this flag will *not* be set.

ISUP_NCI_OG_ECHO_CONTROL_DEVICE

Initially this flag will not be set. When echo suppression is included in the implementation, this flag will be set to indicate that an outgoing echo control device is included in the circuit.

ISUP_FCI_INTERNATIONAL_CALL

For the trial, all calls will be national and this flag will *not* be set.

ISUP_FCI_PASS_ALONG_E2E_METHOD_AVAILABLE**ISUP_FCI_SCCP_E2E_METHOD_AVAILABLE**

Neither of these flags will be set.

ISUP_FCI_INTERWORKING_ENCOUNTERED

This flag is set to indicate that interworking has been encountered on the call (the call is not SS7 end-to-end). In US domestic networks this normally means that an MF or analog trunk or non-SS7 trunk is in the call path.

ISUP_FCI_E2E_INFORMATION_AVAILABLE

This flag will *not* be set.

ISUP_FCI_ISDN_USER_PART_ALL_THE_WAY

This flag will *not* be set.

ISUP_FCI_ORIGINATING_ACCESS_ISDN

This flag will *not* be set.

ISUP_FCI_SCCP_CLNS_METHOD_AVAILABLE**ISUP_FCI_SCCP_CONS_METHOD AVAILABLE****ISUP_FCI_SCCP_ALL_METHODS AVAILABLE**

None of these flags will be set.

ISUP_FCI_NUMBER_TRANSLATED

When the Request-URI contains a *npdi* parameter indicating that the called number has been dippled, this flag will be set.

ISUP_FCI_QUERY_ON_RELEASE

This flag is *not* set.

ISUP_CPC_SUBSCRIBER_ORDINARY

The default calling party's category. When the *cpc* parameter is present in the P-Asserted-Identity of the INVITE, the value of the *cpc* parameter³ will be used to generate the flags with the mapping as follows:

unknown	0	ISUP_CPC_UNKNOWN	
operator	1	ISUP_CPC_OPERATOR_FRENCH	'fr'
operator	2	ISUP_CPC_OPERATOR_ENGLISH	'en'
operator	3	ISUP_CPC_OPERATOR_GERMAN	'de'
operator	4	ISUP_CPC_OPERATOR_RUSSIAN	'ru'
operator	5	ISUP_CPC_OPERATOR_SPANISH	'es'
	6	ISUP_CPC_OPERATOR_LANGUAGE_6	
	7	ISUP_CPC_OPERATOR_LANGUAGE_7	
	8	ISUP_CPC_OPERATOR_LANGUAGE_8	
	9	ISUP_CPC_OPERATOR_CODE_9	

³ See 3GPP TS 29.163 V9.0.0 (2009-12) Annex H.1 and draft-patel-dispatch-cpc-oli-parameter-02.

ordinary	10	ISUP_CPC_SUBSCRIBER_ORDINARY
	11	ISUP_CPC_SUBSCRIBER_PRIORITY
	12	ISUP_CPC_VOICE_BAND_DATA
test	13	ISUP_CPC_TEST_CALL
	14	ISUP_CPC_SPARE
payphone	15	ISUP_CPC_PAYPHONE
mobile-hplmn		
mobile_vplmn		

Note that the **Language** header will be used to identify the language of the operator.

Called Party Number

The *subscriber-number* portion of the *tel* URL or *user-info* portion of the *sip* URL, or the value of the *rn* parameter when present, will be converted to a called party number and included in the *cc_cdpn_length* and *cc_cdpn_offset* fields of the **CC_SETUP_REQ**. The octets of the called party number must conform to the format described in ANSI T1.113.3/3.6. That is: one bit indicating whether the number of address signals is odd (1) or even (0); 7 bits indicating that the number is a national (significant) number (0000011); 3 bits providing the numbering plan indicator is E.164 (001); an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000).

Trunk Group

The *cc_addr_length* and *cc_addr_offset* will contain an **isup_addr** structure. The *scope* field will be set to **ISUP_SCOPE_TG** to indicate that a trunk group is selected; the *id* field will be set to the trunk group number; and the *cic* field will be set to zero (0) to indicate that a circuit is to be automatically selected within the specified trunk group.

Optional Parameters

The *cc_opt_length* and *cc_opt_offset* fields in the **CC_SETUP_REQ** will contain optional parameters encoded according to ISUP formats and codes. This field contains a list of optional parameters consisting of a one-octet parameter tag, a one-octet parameter length, and then a sequence of octets indicated by the parameter length containing the value of the parameter. The list must be terminated with an end-of-optional-parameters parameter.

Optional parameters of a CCI primitive must be formatted precisely as they occur in the optional parameters list of an ISUP message. This optional parameter list consists of a sequence of parameters terminated by an end-of-optional-parameters parameter. Each parameter in the list consists of a single octet specifying the parameter type, followed by another single octet specifying the number of the octets contained in the parameter value, followed by the octet(s) of the parameter value. The end-of-optional-parameters parameter consists of one or two zero octets. When optional parameters are not specified, the optional parameters can be of zero length and no end-of-optional-parameters parameter need appear.

The following optional parameters are populated in the **CC_SETUP_REQ** primitive based on the information available in the **Request-URI** and various headers of the received **INVITE**.

Ported Number

If the *rn* parameter exists in **Request-URI user-info tel** URI portion, the *telephone-subscriber* part of the **tel** URI will be used to populate a *Generic Address Parameter* in the optional parameters. The *Generic Address Parameter* has a tag value of 192; one octet indicating "ported number" (11000000);

one bit odd (1) or even (0) indicator; seven bits indicating national (significant) number (0000011); 3 bits indicating E.164 numbering (001); 2 bits indicating presentation restriction indicator (00); an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000).

Carrier Identification

If the *cic* parameter exists in the **Request-URI tel** URI in accordance with *RFC 4694*, the value of this parameter will be used to populate a *Carrier Identification* parameter in the optional parameters. The *Carrier Identification* parameter has a tag value of 197; 3 bits indicating national network identification (010); 4 bits indicating either a 3-digit (0001) or 4-digit (0010) carrier identification; and 3 or 4 nibble-wide address signals; and, when 3 address signals is provided, a filler (0000).

Carrier Selection

If the *dai* parameter exists in the **Request-URI tel** URI in accordance with *draft-yu-tel-dai-08*, the value of this parameter will be used to populate a *Carrier Selection* parameter in the optional parameters. The *Carrier Selection* parameter has a tag value of 238; and 8 bits indicating the carrier selection. The mapping from values of the *dai* parameter in the **Request-URI** to *Carrier Selection* parameter values is as follows:

no-ind	00000000	no indication (default)
presub	00000001	selected carrier identification presubscribed and not input by calling party
presub-da	00000010	selected carrier identification presubscribed and input by calling party
presub-da-unkwn	00000011	selected carrier identification presubscribed, input by calling party undetermined
da	00000100	selected carrier identification not presubscribed and input by calling party
presub-unkwn-da	00000101	selected carrier identification subscription unknown, input by calling party

Other values of the *dai* parameter described in *draft-yu-tel-dai-08* are not permitted.

Dialed Number

If there is a recognized *telephone-subscriber* portion in the **To** header and it does not correspond to the *Called Party Number* or *Ported Number*, a "dialed number" *Generic Address Parameter* will be added to the optional parameters. The *Generic Address Parameter* has a tag value of 192; one octet indicating "dialed number" (00000000); one bit odd (1) or even (0) indicator; seven bits indicating national (significant) number (0000011); 3 bits indicating E.164 numbering (001); 2 bits indicating presentation restriction indicator - presentation allowed (00); an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000).

Original Called Name

If there is a *display-name* portion in the **To** header and the ISUP CNAM method is marked on the trunk group, an "original called name" *Generic Name* parameter that is presentation-allowed will be added to the optional-parameters. The *Generic Name* parameter has a tag value of 199; two bits indicating presentation restriction set to presentation-allowed (00); one bit indicating availability as available (0); three bits indicating "original called name" (010); and up to 15 IA5 encoded characters. Note that this parameter is not normally used in US networks, but is in Canadian and other networks.

Calling Party's Category

If a *cpc* parameter exists in the **P-Asserted-Identity** header, the value of the calling party parameter in the *cc_call_flags* will be set as indicated previously.

Originating Line Information

If a *oli* parameter exists in the **P-Charge-Info** or **P-Asserted-Identity** header, an *Originating Line Information* parameter will be added to the optional parameters. The *Originating Line Information* parameter has a tag value of 234, and contains 8 bits having the value of the *oli* parameter. Only values in the range from 0x00 to 0x63 are valid (see *ANSI T1.113.3/Annex C*).

Jurisdiction Information

If a *rn* parameter exists in the **P-Charge-Info** or **P-Asserted-Identity** header, a *Jurisdiction Information Parameter* will be added to the optional parameters of the **CC_SETUP_REQ**. The *Jurisdiction Information Parameter* has a tag value of 196 and contains 6 nibble-wide address signals.

Charge Number

If a valid *telephone-subscriber* number is present in the **P-Charge-Info** header, and it is different from the *Calling Party Number* when the *noa* parameter is 1, 2, 3 or absent, or, it is different from the *Called Party Number* when the *noa* parameter is 5, 6, or 7, it will be used in the *Charge Number* parameter in the optional-parameters of the **CC_SETUP_REQ**. When an *npi* or *noa* parameter is also present in the **P-Charge-Info** header, the numbering plan indicator and nature of address in the *Charge Number* will be populated accordingly. This behaviour is consistent with that described in *draft-york-sipping-p-charge-info-07*. The *Charge Number* parameter has a tag value of 235 and contains an odd (1) or even (0) indicator bit; seven bits indicating the nature of address; 1 spare bit; 3 bits indicating the numbering plan as E.164 (001); 4 spare bits; an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000). Only the following nature of address values will be accepted:

- 1 ANI of the calling party, subscriber number
- 2 ANI not available or not provided
- 3 ANI of the calling party, national number
- 5 ANI of the called party, subscriber number
- 6 ANI of the called party, no number present
- 7 ANI of the called party, national number

Calling Party Number

If a valid *telephone-subscriber* number is present in the **P-Asserted-Identity** header, it will be used in the *Calling Party Number* parameter in the optional-parameters of the **CC_SETUP_REQ**. If a **Privacy** header exists and is set to 'id' (or wider scope), the *Calling Party Number* will be marked presentation-restricted; otherwise, it will be marked presentation-allowed. The *Calling Party Number* parameter has a tag value of xxxx and contains a one bit odd (1) or even (0) indicator; seven bits of nature of address indicator coded national (significant) number (0000011); 3 bit numbering plan indicating E.164 (001); two bit address presentation restriction indicator indicating presentation-allowed (00) or presentation-restricted (01); a two bit screening indicator indicating network-provided (11); an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000).

Calling Name

If a *display-name* exists in the **P-Asserted-Identity** header and the trunk is configured for ISUP CNAM method, the *display-name* will be placed in a calling name *Generic Name* parameter in the optional-parameters of the **CC_SETUP_CREQ**. If a **Privacy** header exists and is set to 'user' (or wider scope), the *Generic Name* will be marked presentation-restricted; otherwise, it will be marked presentation-allowed. The *Generic Name* parameter has a tag value of 199; two bits indicating presentation restriction set to presentation-allowed (00) or presentation-restricted (01); one bit indicating availability as available (0); three bits indicating "calling name" (001); and up to 15 IA5 encoded characters.

If there is a *display-name* portion in the **From** header, and the outgoing trunk is marked for ISUP CNAM, a "calling name" *Generic Name* parameter will be placed in the optional-parameters of the **CC_SETUP_REQ** and marked presentation-allowed (if one is not already present). The *Generic Name* parameter has a tag value of 199; two bits indicating presentation restriction set to presentation-allowed (00); one bit indicating availability as available (0); three bits indicating "calling name" (001); and up to 15 IA5 encoded characters.

Note that this parameter is not normally used in US networks, but is in Canadian and other networks.

Additional User Provided Number

If there is a recognized *telephone-subscriber* portion in the **From** header, and it is different from that of the **P-Asserted-Identity** header, an "additional user provided number - not verified" *Generic Address Parameter* will be added to the optional-parameters of the **CC_SETUP_REQ** with presentation-allowed. The *Generic Address Parameter* has a tag value of 192; one octet indicating "additional user provided number - not verified" (00000011); one bit odd (1) or even (0) indicator; seven bits indicating national (significant) number (0000011); 3 bits indicating E.164 numbering (001); 2 bits indicating presentation restriction indicator - presentation allowed (00); an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000).

End of Optional Parameters

When optional parameters are present in the **CC_SETUP_REQ**, the end of optional parameters parameter (with tag value zero (0)) will be appended to the optional-parameters.

Response

After issuing the **CC_SETUP_REQ** primitive on the ISUP stream, the interworking function must await a **CC_OK_ACK** or **CC_ERROR_ACK** primitive before issuing any other primitives on the stream. When a **CC_OK_ACK** primitive is received, the callback function can return. When a **CC_ERROR_ACK** primitive is received, the interworking function should take the following actions based on the *cc_error_type* field in the primitive:

CCSYSERR A system error occurred. The *cc_unix_error* field will contain either [EFAULT] or [EMSGSIZE], indicating that the primitive was poorly formatted (too short or an offset or length field of the primitive referred to a memory extent outside the range of the primitive). The interworking function should log the error and respond with a **500 Server Internal Error** response to the **INVITE** and ultimately remove the session and transaction information.

CCOUTSTATE

The CCI stream upon which the primitive was issued was not in a correct state to accept the primitive (e.g. the stream is not bound). The interworking function

should log the error and respond with a **500 Server Internal Error** response to the INVITE and ultimately remove the session and transaction information.

- CCNOADDR No appropriate address was provided in the *cc_addr_offset* and *cc_addr_length* fields. The interworking function should log the error and respond with a **500 Server Internal Error** response to the INVITE and ultimately remove the session and transaction information.
- CCBADADDR The address provided in the *cc_addr_offset* and *cc_addr_length* fields was incorrectly formatted or contained illegal information. Barring a bug, this error will normally be returned when the trunk group number specified in the address is incorrect. The interworking function should log the error and respond with a **502 Bad Gateway** response to the INVITE and ultimately remove the session and transaction information.
- CCBADDIGS The digits provided in the *cc_cdpn_offset* and *cc_cdpn_length* field was poorly formatted or contained illegal values. The interworking function should log the error and respond with a **500 Server Internal Error** response to the INVITE and ultimately remove the session and transaction information.
- CCBADOPT The optional parameters provided in the *cc_opt_offset* and *cc_opt_length* fields was poorly formatted or contained illegal values. The interworking function should log the error and respond with a **500 Server Internal Error** response to the INVITE and ultimately remove the session and transaction information.
- CCBADCLR The user reference contained in the *cc_user_ref* field is already engaged in a call. The interworking function should log the error and respond with a **500 Server Internal Error** response to the INVITE and ultimately remove the session and transaction information.

7.2.1.2 Successful Setup of Calls from SIP to PSTN

Successful setup of calls from SIP to PSTN will always begin with the receipt (by the callback function associated with the second ISUP stream) of a **CC_SETUP_CON** primitive.

Processing of **CC_SETUP_CON**

When the interworking function receives a **CC_SETUP_CON** primitive, it will use the *cc_user_ref* field of the primitive to locate the call context (session). The interworking function will assign the ISUP provider chosen *cc_call_ref* field from the primitive and make the call context (session) accessible for lookup using this call reference. The ISUP callback function can then return.

Processing of **CC_PROCEEDING_IND**

When the interworking function receives a **CC_PROCEEDING_IND** primitive, it will use the *cc_call_ref* field of the primitive to locate the call context (session). The **CC_PROCEEDING_IND** primitive received by the interworking function will be mapped into a **183 Session Progress** response message for the transaction corresponding to the received INVITE. The **CC_PROCEEDING_IND** message results from the receipt of the first backward message, an ACM from ISUP, without an indication in the ACM that the subscriber is free.

The *cc_flags* field of the **CC_PROCEEDING_IND** primitives contains one or more of the following flags:

ISUP_BCI_NO_CHARGE
ISUP_BCI_CHARGE

These flags are ignored.

ISUP_BCI_ORDINARY_SUBSCRIBER
ISUP_BCI_PAYPHONE

These flags are ignored.

ISUP_BCI_PASS_ALONG_E2E_METHOD_AVAILABLE
ISUP_BCI_SCCP_E2E_METHOD_AVAILABLE

These flags are ignored.

ISUP_BCI_INTERWORKING_ENCOUNTERED

This flag indicates that interworking was encountered between the *Optranex 248* switch and the destination switch. That is, some leg of the call between the *Optranex 248* and the destination switch does not use SS7 signaling. The SIP-ISUP interworking function must cut the call through in both directions to the calling party.

ISUP_BCI_E2E_INFORMATION_AVAILABLE

This flag is ignored and should not normally be set.

ISUP_BCI_ISDN_USER_PART_ALL_THE_WAY
ISUP_BCI_HOLDING_REQUESTED

This flag is ignored and should not normally be set in US networks.

ISUP_BCI_TERMINATING_ACCESS_ISDN

This flag is ignored.

ISUP_BCI_IC_ECHO_CONTROL_DEVICE
ISUP_BCI_SCCP_CLNS_METHOD_AVAILABLE
ISUP_BCI_SCCP_CONS_METHOD_AVAILABLE
ISUP_BCI_SCCP_ALL_METHODS_AVAILABLE

These flags are ignored.

ISUP_OBCI_INBAND_INFORMATION_AVAILABLE
ISUP_OBCI_CALL_DIVERSION_MAY_OCCUR
ISUP_OBCI_ADDITIONAL_INFO_IN_SEG
ISUP_OBCI_MLPP_USER

The ISUP callback function can then return.

Processing of CC_ALERTING_IND

When the interworking function receives a CC_ALERTING_IND primitive, it will use the *cc.call_ref* field of the primitive to locate the call context (session). The CC_ALERTING_IND primitive received by the interworking function will be mapped into a 180 Ringing response message. The CC_ALERTING_IND message results from the receipt of the first backward message, an ACM from ISUP, with an indication in the ACM that the subscriber is free.

Processing of CC_PROGRESS_IND

When the interworking function receives a CC_PROGRESS_IND primitive, it will use the *cc.call_ref* field of the primitive to locate the call context (session). The CC_PROGRESS_IND primitive received by the interworking function will be mapped into an appropriate 1xx response message according to the *cc.event* field of the primitive as follows:

ISUP_EVNT_ALERTING

Indicates that the called party is being alerted. This event is indicated only if a CC_PROCEEDING_IND primitive has already been received. This event is mapped into a 180 Ringing response to the INVITE in accordance with *RFC 3398/7.2.9*.

ISUP_EVNT_PROGRESS

Indicates that the call is progressing with the specified optional parameters. This event is mapped into a 183 **Session Progress** response to the INVITE in accordance with *RFC 3398/7.2.9*.

ISUP_EVNT_IBI

This event is indicated only by the CC_IBI_IND primitive and will not appear here. If this event appears it must be treated as a CC_IBI_IND primitive (see below).

ISUP_EVNT_CALL_FORWARDED_ON_BUSY

This event indicates that the call has been forwarded on busy and the optional parameters (if any) contain the attributes of the forwarding (e.g., redirecting number, etc.). This event is mapped into a 181 **Call is being forwarded** response to the INVITE in accordance with *RFC 3398/7.2.9*.

ISUP_EVNT_CALL_FORWARDED_ON_NO_ANSWER

This event indicates that the call has been forwarded on no answer and the optional parameters (if any) contain the attributes of the forwarding (e.g., redirecting number, etc.). This event is mapped into a 181 **Call is being forwarded** response to the INVITE in accordance with *RFC 3398/7.2.9*.

ISUP_EVNT_CALL_FORWARDED_UNCONDITIONAL

This event indicates that the call has been forwarded unconditionally and the optional parameters (if any) contain the attributes of the forwarding (e.g., redirecting number, etc.). This event is mapped into a 181 **Call is being forwarded** response to the INVITE in accordance with *RFC 3398/7.2.9*.

When the *cc_flags* field contains the flag ISUP_EVNT_PRESENTATION_RESTRICTED, for all events other than ISUP_EVNT_ALERTING, the event will not be mapped to SIP.

Processing of CC_IBI_IND

When the interworking function receives a CC_IBI_IND primitive, it will use the *cc_call_ref* field of the primitive to locate the call context (session). The CC_IBI_IND primitive received by the interworking function will be mapped into a 183 **Session Progress** response message and will connect early media in the direction from the PSTN to the SIP UAC. If the SIP UAC supports the UPDATE method, early media is established toward the SIP UAC using that method.

Processing of CC_CONNECT_IND

When the interworking function receives a CC_CONNECT_IND primitive, it will use the *cc_call_ref* field of the primitive to locate the call context (session). The CC_CONNECT_IND primitive received by the interworking function will be mapped into a 200 OK response message and will connect media in both directions. If the SIP UAC supports the UPDATE method and early media was established toward the UAC using that method, cutting through media in both directions may require an additional UPDATE message.

7.2.1.3 Unsuccessful Setup of Calls from SIP to PSTN

Call Abandonment

After issuing a CC_SETUP_REQ primitive for the call, and before receiving a CC_CONNECT_IND for the call, the interworking function may receive a CANCEL message for the session. The CANCEL message indicates that the caller has abandoned the call. The CANCEL received at the *sofia-SIP* callback

is associated with client information that identifies the call context (session). After looking up the session, the interworking function issues a `CC_RELEASE_REQ` for the call. The `cc_user_ref` field is populated with the user reference for the call when a `CC_SETUP_CON` has not yet been received for the call; otherwise, the `cc_call_ref` field contains the call reference for the call returned in the corresponding `CC_SETUP_CON`. The `cc_cause` value should be set to `CC_CAUS_NORMAL_CALL_CLEARING`.

Automatic Repeat Attempt

After issuing a `CC_SETUP_REQ` primitive for the call, and before receiving a `CC_SETUP_CON` for the call, the interworking function may receive a `CC_CALL_REATTEMPT_IND` primitive. The `CC_CALL_REATTEMPT_IND` primitive indicates that a call repeat attempt should be made by the interworking function and the reason for the reattempt.

The `CC_CALL_REATTEMPT_IND` primitive contains the `cc_user_ref` value that is the same as the interworking function provided on the corresponding `CC_SETUP_REQ` primitive that initiated the call. The interworking function looks up the call context (session) associated with the corresponding `INVITE` and `CC_SETUP_REQ` using this `cc_user_ref` value.

After receiving the `CC_CALL_REATTEMPT_IND` primitive and looking up the call context (session) using the `cc_user_ref` field in the primitive, the interworking function examines a field in the call context to determine whether an automatic reattempt has already been performed. If no reattempt has been performed, the interworking function marks a reattempt⁴ and issues another (can be identical) `CC_SETUP_REQ` primitive to the ISUP stack.⁵

If a reattempt has already been performed, the interworking function examines the `cc_reason` field of the primitive and provides a response on the *sofia-SIP* transaction referenced from the call context as follows:⁶

ISUP_REATTEMPT_DUAL_SEIZURE

The *Optranex 248* is not the controlling exchange for a two-way trunk group and a dual seizure was detected on the two-way trunk group. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message to the `INVITE`, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value `'Q.850;cause=41;text="temporary failure"'` may be added to the response.

The *Optranex 248* will typically be set up (and will so for trial) with outgoing routes to the PSTN that consist of a one-way outgoing trunk group overflowing to a two-way trunk group. The *tgrp* parameter in the **tel** URI of the **Request-URI** field of the `INVITE` specifies a route (a set of overflow trunk groups) rather than a trunk group. When initiating a `CC_SETUP_REQ` primitive, the interworking function should first specify the trunk group associated with the primary (one-way outgoing) trunk. This trunk group number will be an even number. If a `CC_CALL_REATTEMPT_IND` primitive results with a `cc_reason` of `CC_REATTEMPT_CIRCUIT_BUSY`, the interworking function may reattempt on the overflow (two-way) trunk. This trunk (if it exists) will have a trunk group number that is one greater than the trunk group number of the one-way outgoing group and will therefore be an odd number. Should a `CC_CALL_REATTEMPT_IND` result with a `cc_reason` value of `ISUP_REATTEMPT_DUAL_SEIZURE`, the interworking function

⁴ Additional actions on reattempt can include logging the reattempt and the reason for reattempt, or issuing an SNMP trap notification.

⁵ Cases in which the `CC_SETUP_REQ` primitive for the reattempt might not be the same can be where the `CC_CALL_REATTEMPT_IND` is for a circuit in a two-way overflow trunk group: in this case, the reattempt might be for the primary one-way non-overflow trunk group.

⁶ In addition, the reattempt failure and reason can be logged or an SNMP trap notification issued.

might reattempt on the one-way outgoing trunk group. Dual reattempts on the same trunk group should result in release of the call.

ISUP_REATTEMPT_RESET

The *Optranex 248* attempted a call on a circuit which was subsequently reset (either locally or remotely) before the receipt of any backward message for the call. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message⁷ to the INVITE, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value 'Q.850;cause=41;text="temporary failure"' may be added to the response.

ISUP_REATTEMPT_BLOCKING

The *Optranex 248* attempted a call on a circuit which was subsequently blocked (either locally or remotely) before any backward message was received. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message⁸ to the INVITE, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value 'Q.850;cause=41;text="temporary failure"' may be added to the response.

ISUP_REATTEMPT_T24_TIMEOUT

The *Optranex 248* attempted a call for which continuity testing was required; however, tone was not detected within time T24. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message⁹ to the INVITE, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value 'Q.850;cause=41;text="temporary failure"' may be added to the response.

ISUP_REATTEMPT_UNEXPECTED

The *Optranex 248* attempted a call for which an unexpected message was received before any backward message. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message¹⁰ to the INVITE, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value 'Q.850;cause=41;text="temporary failure"' may be added to the response.

ISUP_REATTEMPT_COT_FAILURE

The *Optranex 248* attempted a call which required continuity testing and the result of the continuity test indicated failure. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message¹¹ to the INVITE, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value 'Q.850;cause=41;text="temporary failure"' may be added to the response.

⁷ Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **500 Server Internal Error**.

⁸ Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **500 Server Internal Error**.

⁹ Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **500 Server Internal Error**.

¹⁰ Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **500 Server Internal Error**.

¹¹ Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **500 Server Internal Error**.

ISUP_REATTEMPT_CIRCUIT_BUSY

This reason is only issued by the CCS provider when the scope was `ISUP_SCOPE_CIC` or a specific circuit was specified on the `CC_SETUP_REQ`. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message¹² to the `INVITE`, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value `'Q.850;cause=34;text="no circuit/channel available"'` may be added to the response.

It is the responsibility of the interworking function to reissue the `CC_SETUP_REQ`. Upon the second reattempt for the same call on the same trunk-group, the interworking function should abandon the call and return an error response **503 Service Unavailable** to the `INVITE`. A **Reason** header with value `'Q.850;cause=41;text="temporary failure"'` or `'Q.850;cause=34;text="no circuit/channel available"'` may be added to the response (depending on the value of the `cc_reason` field).

Premature Call Failure

If, after issuing a `CC_SETUP_REQ` primitive for the call, and before receiving a `CC_SETUP_CON` for the call, the interworking function receives a `CC_CALL_FAILURE_IND` primitive, the interworking function will examine the `cc_reason` field of the primitive and responds as follows:

ISUP_CALL_FAILURE_CIRCUIT_BUSY**CC_CAUS_NO_CCT_AVAILABLE**

This failure cause indicates that the selected trunk group is all-circuits-busy. This applies to outgoing calls only. This reason is only issued by the CCS provider when the scope was `ISUP_SCOPE_TG` and no specific circuit was specified on the `CC_SETUP_REQ`. After the first reattempt, the interworking function should return a **503 Service Unavailable** response message¹³ to the `INVITE`, resulting in the ultimate release of the transaction and the corresponding call context (session). A **Reason** header with value `'Q.850;cause=34;text="no circuit/channel available"'` may be added to the response.

ISUP_CALL_FAILURE_T7_TIMEOUT**CC_CAUS_ADDRESS_INCOMPLETE**

This failure cause indicates that there was no response to the call setup request. This applies to outgoing calls only. The interworking function should return a **484 Address Incomplete** response message to the `INVITE`. The **Reason** header may include `'Q.850;cause=28;text="invalid number format (address incomplete)"'`.

Call Rejection

Whereas automatic repeat and premature call failure result from the local detection of call failure, call rejection results from the release of the call by a transit switch or the controlling switch. Call rejection occurs after issuing a `CC_SETUP_REQ` primitive but before receiving a `CC_SETUP_CON` primitive.

When the interworking function receives a `CC_RELEASE_IND` primitive, it uses the `cc_user_ref` field in the primitive to look up the call context (session). The interworking function examines the `cc_cause` field value in the `CC_RELEASE_IND` primitive, issues a `CC_RELEASE_RES` primitive (if a `CC_RELEASE_`

¹² Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **480 Temporarily Unavailable**.

¹³ Note that *RFC 3398* proscribes **503 Service Unavailable** in this case, whereas *Q.1912.5* proscribes **480 Temporarily Unavailable**.

REQ primitive has not already been issued for the call), awaits a CC_OK_ACK primitive and responds on the SIP side as follows:

Q.850	Description	ITU	IETF
CC_CAUS_UNALLOCATED_NUMBER	<i>unallocated (unassigned) number</i>	404	404
CC_CAUS_NO_ROUTE_TO_TRANSIT_NETWORK	<i>no route to network</i>	500	404
CC_CAUS_NO_ROUTE_TO_DESTINATION	<i>no route to destination</i>	500	404
CC_CAUS_SEND_SPECIAL_INFO_TONE	<i>send special information tone</i>	500	—
CC_CAUS_MISDIALLED_TRUNK_PREFIX	<i>misdialed trunk prefix</i>	404	—
CC_CAUS_PREEMPTION	<i>preemption</i>	500	—
CC_CAUS_PREEMPTION_CCT_RESERVED	<i>preemption-circuit reserved for reuse</i>	500	—
CC_CAUS_NORMAL_CALL_CLEARING	<i>normal call clearing</i>	—	—
CC_CAUS_USER_BUSY	<i>user busy</i>	486	486
CC_CAUS_NO_USER_RESPONDING	<i>no user responding</i>	480	408
CC_CAUS_NO_ANSWER	<i>no answer from the user</i>	480	480
CC_CAUS_SUBSCRIBER_ABSENT	<i>subscriber absent</i>	480	480
CC_CAUS_CALL_REJECTED	<i>call rejected</i>	480	403/603
CC_CAUS_NUMBER_CHANGED	<i>number changed</i>	410	301
CC_CAUS_REDIRECT	<i>redirection to new destination</i>	—	—
CC_CAUS_EXCHANGE_ROUTING_ERROR	<i>exchange routing error</i>	480	410
CC_CAUS_OUT_OF_ORDER	<i>destination out of order</i>	502	502
CC_CAUS_ADDRESS_INCOMPLETE	<i>invalid number format (address incomplete)</i>	484	484
CC_CAUS_FACILITY_REJECTED	<i>facility rejected</i>	500	501
CC_CAUS_NORMAL_UNSPECIFIED	<i>normal unspecified</i>	480	480

Table 7.1: Q.850 Cause Value to SIP Response Mapping - Normal Class

Q.850	Description	ITU	IETF
CC_CAUS_NO_CCT_AVAILABLE	<i>resource unavailable</i>	480	503
CC_CAUS_NO_CCT_AVAILABLE	<i>resource unavailable</i>	486	503
CC_CAUS_NETWORK_OUT_OF_ORDER	<i>network out of order</i>	500	503
CC_CAUS_TEMPORARY_FAILURE	<i>temporary failure</i>	500	503
CC_CAUS_SWITCHING_EQUIP_CONGESTION	<i>switching equipment congestion</i>	500	503
CC_CAUS_ACCESS_INFO_DISCARDED	<i>access information discarded</i>	500	503
CC_CAUS_REQUESTED_CCT_UNAVAILABLE	<i>requested circuit/channel not available</i>	500	503
CC_CAUS_PRECEDENCE_CALL_BLOCKED	<i>precedence call blocked</i>	500	503
CC_CAUS_RESOURCE_UNAVAILABLE	<i>resource unavailable, unspecified</i>	500	503
CC_CAUS_NOT_SUBSCRIBED	<i>requested facility not subscribed</i>	500	–
CC_CAUS_OGC_BARRED_WITHIN_CUG	<i>outgoing calls barred within CUG</i>	500	403
CC_CAUS_ICC_BARRED_WITHIN_CUG	<i>incoming calls barred within CUG</i>	500	403
CC_CAUS_BC_NOT_AUTHORIZED	<i>bearer capability not authorized</i>	500	403
CC_CAUS_BC_NOT_AVAILABLE	<i>bearer capability not presently available</i>	500	503
CC_CAUS_INCONSISTENCY	<i>inconsistency in designated outgoing access information and subscriber class</i>	500	–
CC_CAUS_SERVICE_OPTION_NOT_AVAILABLE	<i>service or option not available, unspecified</i>	500	–

Table 7.2: Q.850 Cause Value to SIP Response Mapping - Resource Unavailable Class

Note that in Table 7.2, for CC_CAUS_NO_CCT_AVAILABLE, ITU-T Q.1912.5 specifies that if call completion to busy subscriber is indicated in the backward call indicators, that the response should be 486 Busy Here instead of 480 Temporarily Unavailable.

Q.850	Description	ITU	IETF
CC_CAUS_BC_NOT_IMPLEMENTED	<i>bearer capability not implemented</i>	500	488
CC_CAUS_FACILITY_NOT_IMPLEMENTED	<i>requested facility not implemented</i>	500	488
CC_CAUS_RESTRICTED_BC_ONLY	<i>only restricted digital information available</i>	500	488
CC_CAUS_SERVICE_OPTION_NOT_IMPLEMENTED	<i>service or option not implemented</i>	500	501
CC_CAUS_USER_NOT_MEMBER_OF_CUG	<i>user not member of CUG</i>	500	403
CC_CAUS_INCOMPATIBLE_DESTINATION	<i>incompatible destination</i>	500	503
CC_CAUS_NON_EXISTENT_CUG	<i>non-existent CUG</i>	500	—
CC_CAUS_INVALID_TRANSIT_NTWK_SELECTION	<i>invalid transit network selection</i>	404	—
CC_CAUS_INVALID_MESSAGE	<i>invalid message, unspecified</i>	500	—
CC_CAUS_MESSAGE_TYPE_NOT_IMPLEMENTED	<i>message type non-existent or not implemented</i>	500	—
CC_CAUS_PARAMETER_NOT_IMPLEMENTED	<i>information element/parameter non-existent or not implemented</i>	500	—
CC_CAUS_RECOVERY_ON_TIMER_EXPIRY	<i>recovery on timer expiry</i>	480	504
CC_CAUS_PARAMETER_PASSED_ON	<i>parameter non-existent or not implemented, passed on</i>	500	—
CC_CAUS_MESSAGE_DISCARDED	<i>message with unrecognized parameter, discarded</i>	500	—
CC_CAUS_PROTOCOL_ERROR	<i>protocol error, unspecified</i>	500	500
CC_CAUS_INTERWORKING	<i>interworking unspecified</i>	480	500

Table 7.3: Q.850 Cause Value to SIP Response Mapping - Service or Option Not Available Class

ANSI	Description	SIP
CC_CAUS_UNALLOCATED_DEST_NUMBER	<i>unallocated destination number</i>	404
CC_CAUS_UNKNOWN_BUSINESS_GROUP	<i>unknown business group</i>	404
CC_CAUS_EXCHANGE_ROUTING_ERROR	<i>exchange routing error</i>	480
CC_CAUS_MISROUTED_CALL_TO_PORTED_NUMBER	<i>misrouted call to a ported number</i>	500
CC_CAUS_LNP_QOR_NUMBER_NOT_FOUND	<i>number portability Query on Release (QoR) number not found</i>	500

Table 7.4: ANSI Cause Value to SIP Response Mapping - Normal Class

ANSI	Description	SIP
CC_CAUS_RESOURCE_PREEMPTION	<i>preemption</i>	500
CC_CAUS_PRECEDENCE_CALL_BLOCKED	<i>precedence call blocked</i>	500

Table 7.5: ANSI Cause Value to SIP Response Mapping - Resource Unavailable Class

ANSI	Description	SIP
CC_CAUS_CALL_TYPE_INCOMPATIBLE	<i>call type incompatible with service request</i>	500
CC_CAUS_GROUP_RESTRICTIONS	<i>call blocked due to group restrictions</i>	500

Table 7.6: ANSI Cause Value to SIP Response Mapping - Service or Option Not Available Class

Call Failure

If after issuing a CC_SETUP_REQ primitive and receiving a CC_SETUP_CON primitive for the call, the interworking function receives a CC_CALL_FAILURE_IND primitive, the interworking function will examine the *cc_reason* or *cc_cause* field of the primitive and respond as follows:

ISUP_CALL_FAILURE_RESET

CC_CAUS_NORMAL_UNSPECIFIED

This failure cause indicates that a circuit reset was received for the call during call setup or establishment. This applies to outgoing calls only. The interworking function should return a 480 Temporarily Unavailable response message to the INVITE. The Reason header may include 'Q.850;cause=31;text="normal unspecified"'.

ISUP_CALL_FAILURE_T3_TIMEOUT

ISUP_CALL_FAILURE_OVERLOAD

CC_CAUS_SWITCHING_EQUIP_CONGESTION

This failure cause indicates that the non-controlling exchange received an overload message or automatic congestion control signal that is applied to this call. This applies to outgoing calls only. The interworking function should return a 500 Server Internal Error of 503 Service Unavailable response message to the INVITE. The Reason header may include 'Q.850;cause=42;text="switching equipment congestion"'.

ISUP_CALL_FAILURE_T9_TIMEOUT

CC_CAUS_NO_ANSWER

This failure cause indicates that the call failed while waiting for the distant end to answer. This applies to outgoing calls only.

7.2.2 Interworking Calls from PSTN to SIP

All calls from the PSTN to SIP are initiated upon the receipt of an IAM message the ISUP stack. The interworking function establishes a transaction and a call context. CCI primitives sent to and received from the ISUP stack will be initially received on the first ISUP stream, accepted on for the third ISUP stream, and issued and received subsequently on the third ISUP. CCI primitives issued to the ISUP stack will be issued from within the *nta* message callbacks. CCI primitives will be received on the installed callbacks for the first and third ISUP streams.

7.2.2.1 Mapping of ISUP IAM to SIP INVITE

Processing of CC_SETUP_IND

Upon receiving an IAM message (CC_SETUP_IND primitive) at the ISUP stack beginning a call, the interworking function establishes an *nta* transaction and prepares an INVITE message. The interworking function establishes a call context (session) and assigns the *cc_call_ref* supplied in the CC_SETUP_IND primitive corresponding to the IAM message to identify the call with the ISUP stack.

The interworking function examines the *cc-call-type* field of the `CC_SETUP_IND` primitive. The accepted call types are as follows:

`CC_CALL_TYPE_SPEECH`

This call type represents a speech call.

`CC_CALL_TYPE_3_1kHz_AUDIO`

This call type represents a 3.1 kHz audio call.

`CC_CALL_TYPE_UDI_TA`

This call type represents an unrestricted digital information with tones and announcements call.

For all other call types, the interworking function rejects the call with a `CC_REJECT_REQ` primitive with the cause value (*cc-cause*) of `CC_CAUS_BC_NOT_IMPLEMENTED`.

The interworking function then examines the *cc-call-flags* field of the `CC_SETUP_IND` primitive. The flags are interpreted as follows:

`ISUP_NCI_ONE_SATELLITE_CCT`

`ISUP_NCI_TWO_SATELLITE_CCT`

These flags are ignored.

`ISUP_NCI_CONT_CHECK_REQUIRED`

`ISUP_NCI_CONT_CHECK_PREVIOUS`

These flags can be ignored by call control and are, instead, processed by a separate maintenance application.

`ISUP_NCI_OG_ECHO_CONTROL_DEVICE`

This flag affects the application of echo control on the media circuit. When this flag is set, echo control on the media circuit is not necessary. This flag is seldom set in US domestic networks.

`ISUP_FCI_INTERNATIONAL_CALL`

This flag is ignored for incoming calls and should not typically be set on egress calls.

`ISUP_FCI_PASS_ALONG_E2E_METHOD_AVAILABLE`

`ISUP_FCI_SCCP_E2E_METHOD_AVAILABLE`

These flags can be ignored.

`ISUP_FCI_INTERWORKING_ENCOUNTED`

This flag indicates that interworking was encountered in the course of routing the call to this office. In US domestic networks this normally means that an MF or analog trunk is in the call path and that rather than releasing calls with cause, the playing of in-band tones and announcements will be required in some circumstances. The fact that interworking has been encountered should be saved in the call context (session) data associated with the call as it will affect behaviour later in the call.

`ISUP_FCI_E2E_INFORMATION_AVAILABLE`

This flag is ignored.

`ISUP_FCI_ISDN_USER_PART_ALL_THE_WAY`

This flag is ignored.

`ISUP_FCI_ISDN_USER_PART_NOT_REQUIRED`

This flag is ignored.

ISUP_FCI_ISDN_USER_PART_REQUIRED

If this flag is set, the call is rejected using a **CC_REJECT_REQ** primitive. The *cc-cause* field should be set to **CC_CAUS_FACILITY_REJECTED**.

ISUP_FCI_ORIGINATING_ACCESS_ISDN

This flag is ignored.

ISUP_FCI_SCCP_CLNS_METHOD_AVAILABLE**ISUP_FCI_SCCP_CONS_METHOD_AVAILABLE**

These flags are ignored.

ISUP_FCI_NUMBER_TRANSLATED

When this flag is present, the *npdi* parameter will be added to the *telephone-subscriber* portion of the **tel** URI contained in the **Request-URI** of the resulting INVITE message.

Calling Party's Category

The calling party's category is contained in the *cc.flags* field. The field is first masked (logical AND) with the **ISUP_CPC_MASK** value and then compared to the following values:

ISUP_CPC_UNKNOWN

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to **unknown**, or the *cpc* parameter is omitted from the **tel** URI.

ISUP_CPC_OPERATOR_FRENCH

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to **operator** and the **Language** header is set to **fr**.

ISUP_CPC_OPERATOR_ENGLISH

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to **operator** and the **Language** header is set to **en**.

ISUP_CPC_OPERATOR_GERMAN

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to **operator** and the **Language** header is set to **de**.

ISUP_CPC_OPERATOR_RUSSIAN

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to **operator** and the **Language** header is set to **ru**.

ISUP_CPC_OPERATOR_SPANISH

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to **operator** and the **Language** header is set to **es**.

ISUP_CPC_OPERATOR_LANGUAGE_6

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to 6.

ISUP_CPC_OPERATOR_LANGUAGE_7

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to 7.

ISUP_CPC_OPERATOR_LANGUAGE_8

The *cpc* parameter in the **tel** URI portion of the P-Asserted-Identity header is set to 8.

ISUP_CPC_OPERATOR_CODE_9

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to 9.

ISUP_CPC_SUBSCRIBER_ORDINARY

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to **ordinary**.

ISUP_CPC_SUBSCRIBER_PRIORITY

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to 11.

ISUP_CPC_VOICE_BAND_DATA

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to 12.

ISUP_CPC_TEST_CALL

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to **test**.

ISUP_CPC_SPARE

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to 14.

ISUP_CPC_PAYPHONE

The *cpc* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to **payphone**.

Called Party Number

The *cc_cdpn_offset* and *cc_cdpn_length* fields of the *CC_SETUP_IND* indicate the called party number. This number is extracted from the field and used in the *telephone-subscriber* portion of the **tel** URI in both the **To** header and **Request-URI** of the **INVITE** message.

The called party number field is encoded in the format described in *ANSI T1.113.3/3.6*; that is: one bit indicating whether the number of address signals is odd (1) or even (0); 7 bits indicating that the nature of address; 3 bits providing the numbering plan indicator; an even or odd number of nibble-wide address signals; and, when the number of address signals is odd, a filler (0000). The numbering plan indicator should typically be E.164 (001). The nature of address should typically be national (significant) number (0000011) or international number. When the number is national, a '+1' should be prefixed to the number for the trial. When the number is international, a '+' is prefixed to the number for the trial before placing the number into the *telephone-subscriber* portion of the **tel** URI.

Trunk Group

The *cc_addr_length* and *cc_addr_offset* will contain an **isup_addr** structure. The *scope* field will be set to **ISUP_SCOPE_TG** to indicate that a trunk group was selected; the *id* field will be set to the trunk group number; and the *cic* field will be set to the circuit identification code of the incoming circuit. The *tgrp* parameter in the **tel** URI portion of the **P-Asserted-Identity** header is set to the numeric value of the trunk group number. The circuit identification code is used in conjunction with the signaling point code to which the stream was bound to identify the circuit for media connections.

Optional Parameters

The *cc_opt_length* and *cc_opt_offset* fields of the **CC_SETUP_IND** will contain optional parameters encoded according to ISUP formats and codes. This field contains a list of optional parameters consisting of a one-octet parameter tag, a one-octet parameter length, and then a sequence of octets indicated by the parameter length containing the value of the parameter. The list is terminated with an end-of-optional-parameters parameter. If the field does not follow this format, the setup should be rejected with a **CC_REJECT_REQ** primitive with a *cc_cause* value of **CC_CAUS_INVALID_MESSAGE**.

Optional parameters of the CCI primitive are formatted precisely as they occurred in the optional parameters list of the corresponding ISUP message. This optional parameter list consists of a sequence of parameters terminated by an end-of-optional-parameters parameter. Each parameter in the list consists of a single octet specifying the parameter type, followed by another single octet specifying the number of octets contained in the parameter value, followed by the octet(s) of the parameter value. The end-of-optional-parameters parameter consists of one or two octets. When optional parameters are not specified, the optional parameters can be of zero length and no end-of-optional-parameters parameter need appear.

The following optional parameters may be populated in the **CC_SETUP_IND** primitive that need to be mapped into the information placed into the **Request-URI** and various headers of the **INVITE** being prepared.

Ported Number

If a *Generic Address Parameter* exists in the optional parameters with indication "ported number", then this number will be used as the *telephone-subscriber* portion of the **tel** URI in the **To** header and **Request-URI**. When the parameter is presentation-restricted, it will only be added to the **Request-URI**. The called party number which was overwritten will be added, instead, to an *rn* parameter of the **tel** URI in the **Request-URI** only.

Dialed Number

If a *Generic Address Parameter* is present in the optional parameters with indication "dialed number" and is marked presentation allowed, the number will be used as the *telephone-subscriber* portion of the **tel** URI contained in the **To** header.

Original Called Name

If a *Generic Name* parameter exists in the optional parameters list with indicating "original called name", and the name is set to presentation-allowed, the name will be used as the *display-name* portion of the **To** header.

Originating Line Information

If an *Originating Line Information* parameter is present in the optional parameters, the value of the parameter will be added to an *oli* parameter of the *telephone-subscriber* portion of the **tel** URI contained in the **P-Asserted-Identity** header as well as the **P-Charge-Info** header (if present for other reasons).

Jurisdiction Information

If a *Jurisdiction Information* parameter is present in the optional parameters, the digits from the parameter will be added to an *rn* parameter of the *telephone-subscriber* portion of the **tel** URI contained in the **P-Asserted-Identity** header as well as the **P-Charge-Info** header (if present for other reasons).

Charge Number

If a *Charge Number* parameter is present in the optional parameters a **P-Charge-Info** header will be added to the INVITE message in accordance with *draft-york-sipping-p-charge-info-07*. The *noa* value and *telephone-subscriber* number portions of the header will be taken directly from the parameter.

Calling Party Number

If a *Calling Party Number* parameter is present in the optional parameters list and it is not "network provided" (11), the parameter will be ignored. When network provided, the calling party number will be used in the *telephone-subscriber* portion of the **tel** URI contained in both the **From** and **P-Asserted-Identity** headers of the resulting INVITE. When marked presentation-restricted, the number will only be placed in the **P-Asserted-Identity** header and the **Privacy** header will be elevated to **id** privacy.

Calling Name

If a *Generic Name* parameter exists with the indication "calling name" (001) and it is marked presentation-allowed, the name will be placed in the *display-name* portion of the URI in both the **From** and **P-Asserted-Identity** headers. When the name is presentation-restricted, it will only be placed in the **P-Asserted-Identity** header and the **Private** header will be elevated to **user** privacy.

Additional User Provided Number

If a *Generic Address Parameter* exists with the indication "additional user provided number - not verified" (00000011) in the first octet of the parameter value, the remaining octets in the parameter provide the party number with one bit odd (1) or even (0) indicator; seven bits indicating national (significant) number (0000011) or international number; 3 bits indicating E.164 numbering (001); 2 bits indicating presentation restriction - presentation allowed (00); an even or odd number of nibble-wide address signal; and, when the number of address signals is odd, a filler (0000). When this parameter is present and is presentation allowed, it is used as the *telephone-subscriber* portion of the **tel** URI used in the **From** header of the INVITE message, instead of that contained in the calling party number.

End of Optional Parameters

When optional parameters are present in the **CC_SETUP_IND**, the end of optional parameters parameter (with tag value zero (0)) will be appended to the optional parameters.

Response

The interworking function will issue a **CC_SETUP_RES** indicating the ISUP stream upon which the remaining primitives are to be exchanged (the third stream described above) on the listening stream (the first stream described) and will await receive of either a **CC_OK_ACK** or **CC_ERROR_ACK** primitive (priority message) on the listening stream before continuing. When a **CC_ERROR_ACK** is received, the call context should be dropped and the call abandoned. When a **CC_OK_ACK** is received, the interworking function can issue the prepared INVITE message and either return from the callback or process more received messages on the listening stream.

Note that ISUP should treat these trunks terminating on SIP as interworking trunks. When the **CC_SETUP_IND** is issued and the **CC_SETUP_RES** processed successfully, the T11 timer should be started and an **ACM** message generated automatically if the timer expires.

7.2.2.2 Successful Calls from PSTN to SIP

Note that PRACK, UPDATE and 100 Trying messages, and the like, should be processed by the *sofia-SIP* stack without involving the ISUP stack.

Processing 180 Ringing Response

CC_ALERTING_REQ

Processing 183 Session Progress Response

CC_PROGRESS_REQ

Processing 181 Call Is Being Forwarded Response

CC_PROGRESS_REQ

Processing 182 Queued Response

CC_PROGRESS_REQ

Processing 200 OK Response

CC_CONNECT_REQ

7.2.2.3 Unsuccessful Calls from PSTN to SIP

Processing 3xx Responses

Processing 4xx, 5xx and 6xx Responses

Call Abandonment

Call Failure

If, after receiving a CC_SETUP_IND primitive and before sending a CC_SETUP_RES primitive for the call, the interworking function receives a CC_CALL_FAILURE_IND primitive, the interworking function will examine the *cc_reason* or *cc_cause* field of the primitive and respond as follows:

ISUP_CALL_FAILURE_COT_FAILURE

CC_CAUS_TEMPORARY_FAILURE

This failure cause indicates that the continuity check on the circuit failed. This applies to incoming calls only.

ISUP_CALL_FAILURE_RECV_RLC

CC_CAUS_NORMAL_UNSPECIFIED

This failure cause indicates that the circuit was not completely released by the distant end. This applies to incoming calls only.

ISUP_CALL_FAILURE_BLOCKING

CC_CAUS_TEMPORARY_FAILURE

This failure cause indicates that the circuit was blocked by the receipt of a circuit blocking or circuit group blocking message during call setup. This applies to incoming calls only.

ISUP_CALL_FAILURE_T8_TIMEOUT
CC_CAUS_TEMPORARY_FAILURE

This failure cause indicates that the call failed while waiting for a continuity check report from the distant end. This applies to incoming calls only.

ISUP_CALL_FAILURE_T35_TIMEOUT
CC_CAUS_ADDRESS_INCOMPLETE

This failure cause indicates that additional information (digits) were not received from the caller within a sufficient period of time. This applies to incoming calls only and only when *en block* signaling is not used.

7.2.3 Interworking of Established Calls

For established calls, an *nta* transaction is established for the *sofia-SIP* stack, and a provider call reference is established for the ISUP stack. The transaction is identified by its client identifier for the *sofia-SIP* stack and the ISUP call is identified by its provider call reference. These two identifiers refer to the same call context object (structure) in the interworking function. To process events (callbacks) from the *sofia-SIP* stack, the client identifier is used to retrieve the call context. To process primitives received from the ISUP stack, the provider call reference is used to retrieve the call context.

7.2.3.1 Successful Established Calls

Established calls can be suspended and resumed from either a network or user perspective. Suspend and resume indications from ISUP as indicated by the CC_SUSPEND_IND and CC_RESUME_IND primitives are valid in the CC_CONNECTED or CC_SUSPENDED states.

Processing of CC_SUSPEND_IND

Upon receipt of a CC_SUSPEND_IND, the *cc.call.ref* field of the primitive will be used to retrieve the call context (session). The CC_SUSPEND_IND primitive will be responded to with a CC_SUSPEND_RES primitive and will *not* initially invoke the UPDATE method for the trial. The interworking function will mark the call context (session) as being in the (network or user) suspended state, CCS_SUSPENDED. At a later date, the CC_SUSPEND_IND primitive might invoke the UPDATE or re-INVITE method to suppress transfer of media during the suspend.¹⁴

Processing of CC_RESUME_IND

Upon receipt of a CC_RESUME_IND, the *cc.call.ref* field of the primitive will be used to retrieve the call context (session). The CC_RESUME_IND primitive will be responded to with a CC_RESUME_RES primitive and will *not* initially invoke the UPDATE method for the trial. The interworking function will mark the call context (session) as being in the (network or user) suspended or established state, CCS_SUSPENDED or CCS_CONNECTED. At a later date, the CC_RESUME_IND primitive might invoke the UPDATE or re-INVITE method to restore transfer of media after resume from the suspend.

7.2.3.2 Failed Established Calls

Established calls can fail due to the expiration of a timer within ISUP (in accordance with normal ISUP procedures) or the expiration of a timer within SIP (in accordance with session timers).

¹⁴ The interworking of suspend to media suppression using the re-INVITE method is detailed in *RFC 3398 Section 9*.

Processing of CC_CALL_FAILURE_IND

ISUP_CALL_FAILURE_T2_TIMEOUT

CC_CAUS_NORMAL_CALL_CLEARING

This failure cause indicates that the controlling exchange received a user suspend message but did not receive a user resume message within time T2. This applies to all established calls.

ISUP_CALL_FAILURE_T6_TIMEOUT

CC_CAUS_NORMAL_CALL_CLEARING

This failure cause indicates that the non-controlling exchange received a network suspend message but did not receive a network resume message within time T6. This applies to all established calls.

ISUP_CALL_FAILURE_T38_TIMEOUT

CC_CAUS_RECOVERY_ON_TIMER_EXPIRY

This failure cause indicates that the call was (network) suspended beyond the allowable period. This applies to all established calls.

7.2.4 Interworking of Call Release

7.2.4.1 Calls Released by the SIP Network

Processing of CANCEL

Processing of BYE

7.2.4.2 Calls Released by the ISUP Network

Release by the Called Party

Release by the Calling Party

Processing of CC_RELEASE_IND

When the interworking function receives a CC_RELEASE_IND primitive for an established call, it uses the *cc_call_ref* field in the primitive to look up the call context (session). The interworking function examines the *cc_cause* field value in the CC_RELEASE_IND primitive, issues a CC_RELEASE_RES primitive (if a CC_RELEASE_REQ primitive has not already been issued for the call), awaits a CC_OK_ACK primitive and responds on the SIP side as illustrated in [Table 7.1](#), [Table 7.2](#), [Table 7.3](#), [Table 7.4](#), [Table 7.5](#) and [Table 7.6](#). However, if the cause value (*cc_cause*) is CC_CAUS_NORMAL_CALL_CLEARING, a BYE message is sent instead of a 4xx, 5xx or 6xx response.

7.2.5 SDP Handling

The *OpenSS7 Media* stack is illustrated in [Figure 7.1](#) and interactions between the interworking function and the *OpenSS7 Media* stack are illustrated in [Figure 7.3](#).

The *OpenSS7 Media* stack and *Media Gateway Interface (MGI)* are used for interworking TDM media connections with RTP. The *Optranex 248* uses the *OpenSS7* media stack for processing media streams. The */dev/mg* driver is used for interworking TDM media streams into RTP media streams.

A purpose-developed interworking function exists at the control nexus between the *sofia-SIP* and *OpenSS7 Media* stacks.

7.3 Carrier Proxy

The protocol architecture described in [Chapter 6 \[Protocol Architecture\]](#), [page 37](#) requires the implementation of a carrier proxy that communicates between customer proxies, the *C-SCP*, and the *Optranex 248* gateway switch.

8 Hardware Architecture

The hardware of the *Optranex 248* switch is based on two *Kontron* hardened (NEBS 3 compliant) PC server chassis, each equipped with the *Optranex 248* SONET/SDH card.

8.1 Power and Cooling

The hardened (NEBS 3 compliant) PC servers are available with either 110/220VAC or -48VDC single or dual redundant power supplies. Redundant DC power supplies are required for the Central Office environment. The -48VDC power supplies are rated at 450 Watts per server. In a two-server configuration, a maximum of 900 Watts of power (approximately 20 Amps at -48VDC) is required. Cooling for a maximum of 900 Watts of power dissipation is required.

8.1.1 Grounding

The DC chassis provides two #10-32 threaded studs for chassis grounding. A single 90-degree standard barrel, two-hole, compression terminal lug with 5/8-inch pitch suitable for #14-10 AWG conductor (such as the Thomas & Betts' terminal lug P/N 256-31426-141) must be used for proper safety grounding. A crimping tool may be needed to secure the terminal lug to the grounding cable.

8.2 Mounting

Mounting rails are available for a standard 19" dual or quad rail rack. Mounting rails for the Dallas InfoMart will require specification of whether the colo is in dual or quad rail 19" racks.

8.3 Ethernet Network Interfaces

Initial requirements are for 200 Mbps full-duplex capacity between the *Optranex 248* switch servers and the customer proxy and SIP/RTP network. For load testing, however, 4 Gbps full-duplex capacity between the *Optranex 248* switch servers and the customer proxy or other SIP/RTP network will be required.

Each server in the two-server configuration has four GigE NIC ports. These ports are accessible from the rear of the chassis. There are no GigE NIC ports accessible from the front of the chassis. *The GigE NIC ports are intended to be installed with shielded cabling that is grounded at both ends of the cable.*

The GigE NIC ports are intra-building ports suitable for connection to intra-building or unexposed wiring or cabling only. These intra-building ports *must not* be connected to, nor come into metallic contact with, interfaces that connect to the outside plant or its wiring. These interfaces are designed for use as intra-building interfaces only (Type 2 or 4 ports as described in GR-1089-CORE, Issue 4) and require isolation from the exposed outside plant cabling. Note that the addition of primary protectors is *not* sufficient protection for these interfaces to connect to, or come into metallic contact with, outside plant wiring.

8.4 SONET/SDH Network Interfaces

Initial requirements are for OC-12 1:1 APS connections from the PSTN (SONET ADM equipment). This OC-12 pipe will support a maximum of 12 DS-3s or 8064 channels; however, trunks are only allocated for an aggregate DS-3 capacity of 3 x DS-3s or 2016 channels.

The platform is capable of 2 x OC-48 1:1 APS connections (maximum of 96 DS-3s or 64,512 channels) or 4 x OC-48 unprotected connections (maximum of 192 DS-3s or 129,024 channels).

8.4.1 Small Formfactor Pluggable Modules

For trial, two SFP pluggable modules will be provided capable of OC-12 Single-Mode Fiber Short Reach (1310 nanometer) connectivity using two LC connectorized fibers per module. One SFP module will be plugged into the *Optranex 248* card installed in each of the two hardened (NEBS-3 compliant) PC servers. The SFP modules will be configured for 1:1 APS operation. These modules will be Finisar FTLF1322P1BTR (LC duplex, OC-12/3, SR, SF, 1310 nm) or equivalent.

8.4.1.1 Finisar Modules

Finisar has been providing SFP modules to the telecommunications industry for some time and is one of the market leaders in these devices. Finisar SFP modules are available from a number of popular suppliers including **Digi-key**, **Avnet** and **Nu Horizons**.

Suitable Finisar SFP modules are listed below in [Table 8.1](#).¹

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
FTLF1321P1BTL	OC-48/12/3	S	1310	5	775-1016-ND	\$164.34	
FTLF1521P1BCL	OC-48/12/3	S	1310	15	775-1030-ND	\$464.40	
FTLF1421P1BCL	OC-48/12/3	S	1310	20	775-1023-ND	\$188.50	
FTLF1721P2BCL	OC-48/12/3	S	1310	45	775-1035-ND	\$464.40	\$529.23
FTRJ1721P1BCL	OC-48/12/3	S	1310	45			\$509.23
FTLF1621P2BCL	OC-48/12/3	S	1550	80	775-1034-ND	\$642.60	\$732.31
FTRJ1621P1BCL	OC-48/12/3	S	1550	80			\$692.31
FTLF1322P1BTR	OC-12/3	S	1310	15	775-1019-ND	\$86.23	\$80.00
FTRJ1422P1BTL	OC-12	S	1310	40	775-1024-ND	\$172.55	\$193.85
FTLF1522P1BTL	OC-12	S	1550	80	775-1031-ND	\$249.40	
FTLF1217P2BTL	OC-3/1	M	1310	2	775-1011-ND	\$58.00	\$52.73
FTLF1323P1BTR	OC-3	S	1310	15		\$87.40	\$70.77
FTLF1323P1BTL	OC-3	S	1310	40	775-1018-ND	\$99.60	
FTLF1523P1BTL	OC-3	S	1550	80	775-1032-ND	\$230.55	\$244.62

Table 8.1: *Finisar SFP Modules*

Most of these Finisar SFP modules are available from **Digi-key**, **Avnet** and **Nu Horizons**.

For intra-office applications scaling to a maximum OC-3 rate, the FTLF1217P2BTL is the best value (\$52.73); OC-12/3 rate, FTLF1322P1BTR (\$80.00); OC-48/12/3 rate, FTLF1321P1BTL (\$164.34).

Two FTLF1217P2BTL (2 x \$80.00) and two FTLF1321P1BTL (2 x \$164.34) SFP modules installed an *Optranex 248* card can provide every possible configuration of the card: from 4 x OC-3 through 2 x OC-48, under software control.

8.4.1.2 Avago Technologies Modules

Suitable Avago Technologies SFP modules are listed below in [Table 8.2](#).

¹ Note that prices shown are approximate.

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet	DMI
AFCT-5745PZ	OC-48/12/3	S	1310	2			\$223.08	Y
AFCT-5745TPZ	OC-48/12/3	S	1310	15			\$312.31	Y
AFCT-5755LZ	OC-12/3	S	1310	2			\$87.69	Y
AFCT-5755PZ	OC-12/3	S	1310	2			\$87.69	Y
AFCT-5755TLZ	OC-12/3	S	1310	15			\$106.92	Y
AFCT-5755TPZ	OC-12/3	S	1310	15			\$106.92	Y
AFCT-5765LZ	OC-3	S	1310	2			\$73.85	Y
AFCT-5765PZ	OC-3	S	1310	2			\$73.85	Y
AFCT-5765TLZ	OC-3	S	1310	15			\$87.69	Y
AFCT-5765TPZ	OC-3	S	1310	15			\$87.69	Y

Table 8.2: *Avago SFP Modules*

Avago provides a number of related modules; however, these variations either do not provide DMI (without reducing cost), or provide a temperature wider than the operating temperature range of the *Optranex 248* at additional and unnecessary cost.

These Avago SFP modules are not available from [Digi-key](#) but are available from [Avent](#). As an option, the AFCT-5765PZ for systems scaling to OC-3; AFCT-5755PZ (\$73.85), OC-12/3 (\$87.69); AFCT-5745PZ (\$223.08), OC-48/12/3; may be used instead of the Finisar equivalents.

8.4.1.3 JDSU SFP Modules

Suitable [JDSU](#) SFP modules are listed below in [Table 8.3](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Nuhorizons
CT2-MS1LBT33C5	OC-48/12/3	S	1310	2			\$232.70
CT2-MS1LKT33C5	OC-48/12/3	S	1310	2			\$240.50
CT2-MI1LBT33C5	OC-48/12/3	S	1310	15			\$302.90
CT2-MI1LKT33C5	OC-48/12/3	S	1310	15			\$310.70
CT2-MI2LBT53C5	OC-48/12/3	S	1550	20			\$539.50
CT2-MI2LKT53C2	OC-48/12/3	S	1550	20			\$619.89
CT2-ML1LBT31C4	OC-48/12/3	S	1310	40			\$577.20
CT2-ML2LBT51C4	OC-48/12/3	S	1550	80			\$824.50
CT2-ML2LKT51C1	OC-48/12/3	S	1550	80			\$875.52
CT2-ML2LKT51C4	OC-48/12/3	S	1550	80			\$838.50
CT2-KI1MATD31C1	OC-12/3	S	1310	15			\$89.47
CT2-KI1PKTD31C1	OC-12/3	S	1310	15			\$128.70
CT2-PL1LKT33C4	OC-12/3	S	1310	40			\$299.00
CT2-PL2LKT53C4	OC-12/3	S	1550	80			\$318.50
CT2-LI1MATD31C1	OC-3	S	1310	15			\$75.41
CT2-LI1PKTD31C1	OC-3	S	1310	15			\$122.20
CT2-NI1LKT33C2	OC-3	S	1310	15			\$150.80
CT2-NL1LBT33C2	OC-3	S	1310	40			\$195.55
CT2-NL2LBT52C2	OC-3	S	1310	80			\$378.45

Table 8.3: *JDSU SFP Modules*

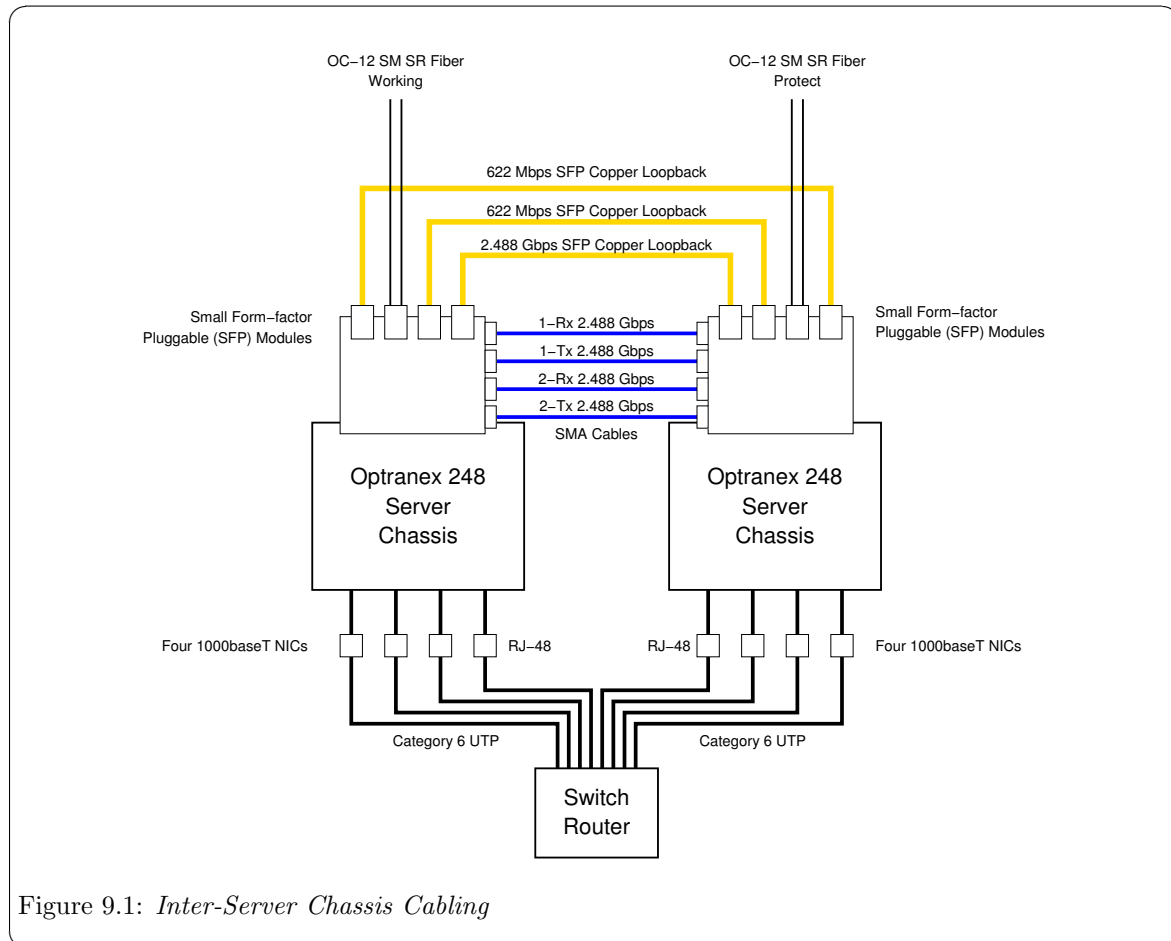
These JSD Uniphase SFP modules are no available from [Digi-key](#) nor [Avnet](#) but are available from [Nu Horizons](#).

Additional modules are listed under [Section A.1 \[Alternate SFP Modules\]](#), page 151.

9 Platform Architecture

9.1 Inter-Server Chassis Cabling

Inter-server chassis cabling is provided for hardware APS connection and loopback between the two servers making up the *Optranex 248* switch, as illustrated in [Figure 9.1](#).



Inter-server chassis cabling for the two hardened PC servers of the *Optranex 248* switch will consist of two components:

1. One set of 4 SMA male connectorized coaxial cables (illustrated in blue in [Figure 9.1](#)).
2. One SFP module connectorized loopback cable (illustrated in gold in [Figure 9.1](#)).
3. Two additional optional SFP module connectorized loopback cables (illustrated in gold in [Figure 9.1](#)).

9.1.1 SMA Cables

The purpose of SMA connectorized coaxial cabling is to provide hardware APS connectivity between the two hardened PC servers that make up the *Optranex 248* switch. One set of four SMA male

connectorized coaxial cables will be provided for hardware based APS interconnection between the two *Optranex 248* cards installed in each of the hardened PC chassis. The SMA cables will carry receive and transmit paths for each of 2 x 2.5 Gbps AC-coupled LVDS hardware APS connections. The 2 x 2.5 Gbps AC-coupled LVDS connections provide connection for hardware-based APS failover between the chassis. In this configuration, the each *Optranex 248* server is capable of connecting to either SFP OC-12 module installed in separate chassis.

9.1.2 SFP Loop-back Cables

The purpose of the SFP loop-back cables is to provide for performance and capacity testing of the platforms. SFP loop-back cables capable of OC-48 speeds (2.488 Gbps) will be provided.¹ These loopback cables plug into one SFP pluggable module interface on each of the two chassis and will be capable of looping back 48 DS-3s worth of traffic (32,256 channels). The loop-back cables are solely for performance and capacity testing of the platforms and will be not be used under normal operation.

Two additional optional SFP copper loopback cables capable of OC-12 or better operation may also be provided to meet specific test objectives during trial.

9.2 Spares

Spares will not be provided for the initial trial. Spare modules consist of the following:

- spare Intel Xeon 5400 series processor;
- spare memory modules;
- spare pluggable 450 Watt -48 VDC power supplies;
- spare pluggable fan modules (CPU fans, IO fans);
- spare filter packs;
- spare pluggable SAS harddrives;
- spare *Optranex 248* SONET/SDH interface PCI Express cards; and,
- spare pluggable OC-12 SFP modules.

Note that with the exception of SONET optical modules, SAS harddrives and power supplies, replacement of parts including processors, memory modules, fan modules, filter packs, and PCI cards requires powering down at least one of the redundant servers. SONET optical modules, SAS harddrives and power supplies are hot-pluggable.

¹ Cable Molex SD-74742-001 (Digi-key PN WM1666-ND) or equivalent.

10 Management Architecture

10.1 Local Alarms

Each of the servers in the *Optranex 248* switch provides a local dry-contact alarms interface.

10.2 Remote Management

Each server in the *Optranex 248* platform requires a Remote Management Module and associated NIC.¹ These remote management modules are responsible for the remote management of the compute platforms. They provide HTTP, SNMP, and IPMI management capabilities.

10.3 SNMP Management

The SNMP management framework for the *Optranex 248* gateway switching application is illustrated in [Figure 7.4](#).

10.3.1 OpenSS7 Management Information Bases

10.3.1.1 STREAMS MIB

The *OpenSS7* management information base used for the management of the STREAMS framework is provided by the `OPENSS7-STREAMS-MIB` and `OPENSS7-STREAMS-EXT-MIB`. An agent implementing the complete MIBs is available for trial.

10.3.1.2 SCTP MIB

`RFC3873-MIB (SCTP-MIB)`

10.3.1.3 M3UA MIB

The *M3UA* and *MTP* management information bases are responsible for the management, measurement, monitoring, auditing, alarming and configuration of the *OpenSS7* SIGTRAN stack. Only the *Application Server* portions of the *M3UA MIB* are used for this application. Only those portions of the *MTP MIB* applicable to SIGTRAN M3UA are used for this application.

The *OpenSS7* management information base used for the management of the SIGTRAN stack is provided by the `OPENSS7-M3UA-ASP-MIB`. Note that, for the Dallas trial, this MIB will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIB will be provided for in-service.

10.3.1.4 ISUP MIB

The *OpenSS7* enterprise management information bases used for the management of the ISUP component are contained in the `OPENSS7-ISUP-MIB` and `OPENSS7-ISUP-OM-MIB`. The former is responsible for the configuration, management and alarms associated with the ISUP stack. The later is responsible for operational measurements associated with the ISUP stack.

Note that, for the Dallas trial, these MIBs will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIBs will be provided for in-service.

¹ Intel Remote Management Module 2 (Intel RMM2) and RMM NIC.

10.3.1.5 RTP MIB

Note that, for the Dallas trial, these MIBs will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIBs will be provided for in-service.

10.3.1.6 SONET/SDH MIB

Note that, for the Dallas trial, these MIBs will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIBs will be provided for in-service.

10.3.1.7 MX MIB

The *OpenSS7* enterprise management information bases used for the management of the multiplex components are contained in the `OPENSS7-MX-MIB`, `RFC2495-MIB` (DS1-MIB), `RFC3895-MIB` (DS1-MIB), `RFC4805-MIB` (DS1-MIB), `RFC4805-EXT-MIB` (DS1-EXT-MIB), `RFC2496-MIB` (DS3-MIB) and `RFC3896-MIB` (DS3-MIB).

Note that, for the Dallas trial, these MIBs will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIBs will be provided for in-service.

10.3.1.8 SWITCH MIB

The *OpenSS7* enterprise management information bases used for the management of the switching of channel and multiplex components are contained in the `OPENSS7-MX-MIB`, `OPENSS7-PHY-MIB`, `RFC2494-MIB` (DS0-MIB) and `RFC2494-EXT-MIB` (DS0BUNDLE-MIB).

Note that, for the Dallas trial, these MIBs will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIBs will be provided for in-service.

10.3.1.9 MG MIB

The *OpenSS7* enterprise management information bases used for the management of the media gateway are contained in the `OPENSS7-MG-MIB` and `OPENSS7-MG-OM-MIB`.

Note that, for the Dallas trial, these MIBs will only be implemented to the point necessary to performing configuration and basic alarms necessary for trial purposes. A completed implementation of the MIBs will be provided for in-service.

11 System Architecture

12 Logistics

12.1 Site Preparation

12.1.1 Rack

Racks must be one of the following:

- 19-inch 2-Post rack with EIA Universal and EIA Wide hole spacing, and equipment mounting posts from 3 to 5 inches deep.
- 19-inch 4-Post rack with EIA Universal and EIA Wide hole spacing, and front equipment mounting rail to rear equipment mounting rail separation not exceeding 24 inches.
- 23-inch 2-Post rack with EIA Universal, EIA Wide and ETSI hole spacing, and equipment mounting posts from 3 to 5 inches deep.
- 23-inch 4-Post rack with EIA Universal, EIA Wide and ETSI hole spacing, and front equipment mounting rail to rear equipment mounting rail separation not exceeding 24 inches.

Physical dimensions and minimum clearances are listed in [Table 12.1](#).

Physical Dimensions	inches	mm
Height	1.70	43.2
Width	16.93	430.0
Depth	20.0	508
Clearance (minimum)	inches	mm
Front	2.0	76
Side	1.0	25
Rear	3.6	92

Table 12.1: *Physical Dimensions*

Note that additional clearance (to 6 inches) may be required front and rear to ensure that front inlet temperature does not exceed rated maximums and to ensure that rear exhaust does not interfere with other devices mounted in the same rack. Also, clearances are the minimum required to provide adequate ventilation given nominal environmental conditions. Additional clearance is required to provide access to the equipment for service of hard drives and power supplies.

12.1.1.1 Rack Mounts

A separate rack mount kit is required depending upon the rack. One rack mount kit is required for a 19" 2- or 4-post rack, another for a 23" 2- or 4-post rack. Rack mount kits will be provided with the *Optranex* servers.

12.1.1.2 Cable Management

Rack mount kits for the *Optranex* servers do not include cable management devices. Sufficient slack in cabling, both DC power and interface cables, must be provided to allow each server chassis to be pulled out 24 inches. Otherwise, any cabling or power feeds must be disconnected before sliding out the equipment on its equipment rails.

12.1.2 Power

The hardened (NEBS 3 compliant) PC servers are available with either 110/220VAC or -48VDC single or dual redundant power supplies. Redundant DC power supplies are required for the Central Office environment and are provided for trial and in-service.

12.1.2.1 DC Power

The -48VDC power supplies are rated at 450W per server. Breakers or fuses should be rated at no more than 10 amps (per server). For more details on DC power requirements, see [Section 12.3.2.1 \[DC Power\]](#), page 142. Breaker or fuse panels for trial will be provided by *Heramax*.

Grounding

Each DC server chassis provides to #10-32 threaded studs for chassis grounding. A single 90-degree standard barrel, two-hole, compression terminal lug with 5/8-inch pitch suitable for #14-10 AWG conductor (such as Thomas & Betts' terminal lug P/N 256-31426-141) must be used for proper safety grounding. A crimping tool may be needed to secure the terminal lug to the grounding cable. For more details on grounding, see [\(undefined\) \[\(undefined\)\]](#), page [\(undefined\)](#).

12.1.2.2 AC Power

The *Optranex* servers for the trial are DC powered. AC power will not be required for either the trial or in-service.

12.1.3 Cooling

12.1.3.1 Temperature

The temperature, in which each server operates when installed in an equipment rack must not go below 5 degrees Celcius (41 degrees Fahrenheit) or rise above 35 degrees Celcius (95 degrees Fahrenheit). Extreme fluctuations in temperature can cause a variety of problems.

12.1.3.2 Ventilation

The equipment rack must provide sufficient airflow to the front of each server to maintain proper cooling. The rack must also include ventilation sufficient to exhaust a maximum of 1023 BTUs (British Thermal Units) per hour for each server, or 2046 BTU per hour total. The rack selected and the ventilation provided must be suitable to the environment in which the servers will be used. Physical dimensions and minimum clearances for adequate ventilation are listed in [Table 12.1](#).

12.1.4 Connections

12.1.4.1 Ethernet Connections

Each of the two server chassis has 4 x 1000baseT RJ-45 Ethernet NIC primary interfaces, and 1 x 100baseT RJ-45 Ethernet NIC for the remote management module. This is a total of 8 x 1000baseT RJ-45 Ethernet NIC primary interfaces and 2 x 100baseT RJ-45 Ethernet NIC remote management modules, or, a total of 10 RJ-45 connections for trial. Provision should be made for growth to an additional two server chassis at in-service.

Proper alien noise immunity on all Ethernet NIC cables requires Category 6 or 6a STP (Shielded Twisted Pair). These cables will be provided for trial and in-service by *Heramax* at installation.

For additional information on Ethernet connections, see [Section 4.1 \[Ethernet Connections\]](#), page 22.

12.1.4.2 SONET Connections

Fiber patch cords for connection between GC optical passive cross-connects and the *Optranex* server chassis will be provided by GC or *HeraMax*. Patch cords will be compliant to OC-12/3, short-reach, single-mode, 1310nm, 9/125 fiber. The *Optranex* end of the fiber patch cords must have LC duplex connectors.

Additional loop-back copper module terminated cables will be used for performance testing. These copper loop-back cables will be provided by *Optranex* for trial and are not required for in-service.

For additional information of SONET/SDH connections, see [Section 4.2 \[SONET/SDH Connections\]](#), [page 32](#).

12.1.4.3 Management Connections

Additional management connections, beyond the Ethernet RMM connections which were included above under Ethernet Connections, such as the serial interface and local alarms panel dry contact interface will not be used for trial.

12.1.4.4 Hardware APS Connections

Hardware APS connections use SMA coaxial cables that connect between the server chassis. These cables will be provided by *Optranex* and need to be installed between the interface card SMA connectors. These connections might not be used for trial but will be used for in-service.

12.2 Equipment List

12.2.1 Servers

The *Optranex 248* consists of two server chassis. Each is a Kontron TIGW1U (product code TMRA0201W or TMRD0201W or equivalent) server.

12.2.1.1 Processors

Each server is equipped with two Quad-Core Intel Xeon 5400 or 5500 series processors, preinstalled.

12.2.1.2 Memory

Each server is equipped with DDR3 memory modules, preinstalled.

12.2.1.3 Hard Drives

The *Optranex 248* servers are each equipped with 3 hot-swappable SAS (Serial Attached SCSI) hard-drives, preinstalled.

12.2.1.4 PCIe Riser

Each *Optranex 248* server is equipped with a PCIe Riser Card inserted into a PCI Super Slot. The PCIe Riser card provides for x1, x4 or x8 lane operation. The *Optranex 248* SONET/SDH cards are x4 lane PCIe cards. The *Optranex 192* SONET/SDH cards are x8 lane PCIe Cards.

12.2.1.5 Power Supplies

Each *Optranex 248* server can be equipped with one or two (redundant, hot-swappable) AC or DC power supplies. Each server chassis will be provided with two (redundant hot-swappable) DC power supplies, preinstalled.

12.2.2 Interface Cards

Each *Optranex 248* server is equipped with either a *Optranex 248* x4 lane PCIe or *Optranex 192* x8 lane PCIe SONET/SDH card. For the Dallas trial, each server will be equipped with one *Optranex 248* x4 lane PCIe card. Interface cards will be installed separately and after server installation.

12.2.3 SFP Modules

Each *Optranex 248* SONET/SDH card accepts up to four SFP modules. SFP modules must be compliant with *SFF Committee INF-8074i Revision 1.0* specifications. SFP modules will be provided by *Optranex* and will be installed with the interface cards.

12.2.4 Cables

Cable requirements are as follows:

- 10 Category 6 or 6a STP RJ-45 patch cords, 5 between the rear of each *Optranex* server chassis and the Juniper router.
- 2 9/125 fiber patch cords with LC duplex connectors, one between the GC optical patch panel and the rear of each *Optranex* server chassis.
- 4 x SMA coaxial patch cords between the rear of the *Optranex* server chassis.
- 3 x SFP copper patch cables between the rear of the *Optranex* server chassis.

12.2.5 Rack Mount Kits

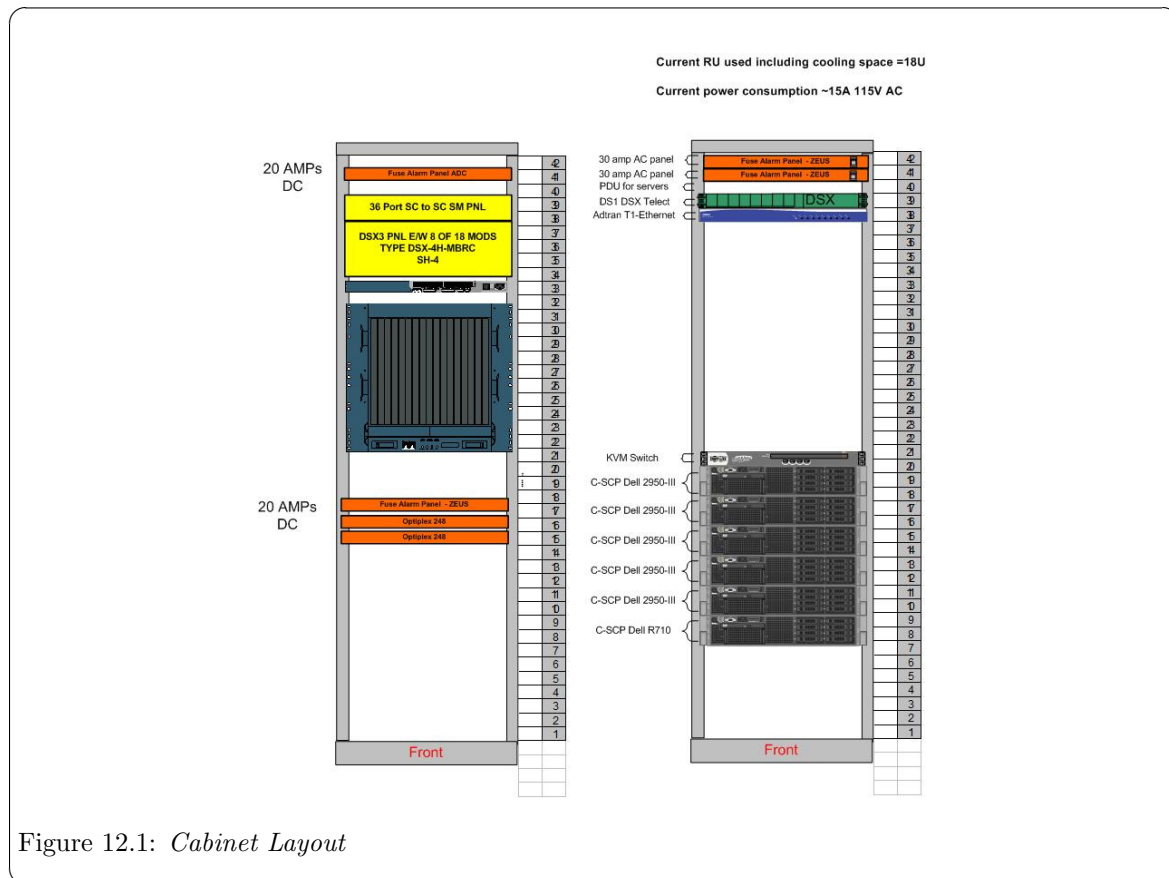
Two 19" rack mount kits will be provided by *Optranex*: one for each server chassis.

12.2.6 Cable Management Kits

The *Optranex* servers must be disconnected from power and interface units before the case is opened for service. Cable managements kits are not required when connections are removed before the server is slid out on its equipment rails.

12.3 Installation

The cabinet layout for the Dallas trial is illustrated in [Figure 12.1](#).

Figure 12.1: *Cabinet Layout*

12.3.1 Rack

Two or more people are required to install the server chassis in the rack. Room air temperature must be below 95 degrees Fahrenheit (35 degrees Celsius) during installation and operation. Air vents (front and back) must have at least 6 inches (15 centimeters) clearance for proper ventilation. Device installation should begin from the bottom of the rack upward. The heaviest devices should be installed toward the bottom of the rack. Not more than one device should be extended out of the rack at a time. Remove rack doors and side panels to provide unrestricted access during installation. Ensure that devices are attached to properly grounded power feeds and that multiple devices do not overload the power outlets during installation. Do not place any object weighing more than 110 pounds (50 kilograms) on top of rack-mounted devices.

Rack installation requires flat-blade and Phillips screwdrivers and rack mount kit appropriate for the rack in which the server chassis is to be installed. There are two rack mount kits: one for 19" 2- and 4-post racks; another for 23" 2- and 4-post racks.

12.3.2 Power

Servers can be installed using one or two (redundant) AC or DC power supplies. For evaluation and laboratory installations, only one AC or DC power supply is required. Requirements for DC and AC power feeds are discussed in [Section 12.3.2.1 \[DC Power\]](#), page 142 and [Section 12.3.2.2 \[AC Power\]](#), page 142.

12.3.2.1 DC Power

Connection with a DC (Direct Current) source should only be performed by trained service personnel. The server with DC input is to be installed in a Restricted Access Location in accordance with articles 110-26 and 110-27 of the National Electric Code, ANSI/NFPA 70. The DC source must be electrically isolated by double or reinforced insulation from any hazardous AC source. The DC source must be capable of providing up to 300 watts of continuous power per feed pair.

Mains DC Power Disconnect

You are responsible for installing a properly rated DC power disconnect for the server system. This mains disconnect must be readily accessible, and it must be labeled as controlling power to the servers. The UL Listed circuit breaker of a centralized DC power system may be used as a disconnect device when easily accessible and should be rated no more than 10 amps.

Grounding the Server

The servers are intended for installation with an isolated DC return (DC-I) and is to be installed in a Common Bonding Network (CBN) per NEBS GR-1089. To avoid the potential for an electrical shock hazard, the servers must be reliably connected to an earth grounding conductor. The earth grounding conductor must be a minimum 14AWG connected to the earth ground studs on the rear of each server. The safety ground connector should be connected to each chassis stud with a Listed closed two-hole crimp terminal having a 5/8-inch pitch. The nuts on the chassis earth ground studs should be installed with a 10 in-lb torque. The safety ground conductor provides proper grounding only for each server chassis connected. Additional, proper grounding must be provided for the rack and other devices installed in it.

Each DC chassis provides two #10-32 threaded studs for chassis enclosure grounding. A single 90-degree standard barrel, two-hole, compression terminal lug with 5/8-inch pitch suitable for #14-10 AWG conductor (such as the Thomas & Betts' terminal lug Part Number 256-31426-141) must be used for proper safety grounding.

Overcurrent Protection

Overcurrent protection UL Listed circuit breakers must be provided as part of each host equipment rack and must be incorporated in the field wiring between the DC source and the server. The branch circuit protection shall be rate minimum 75Vdc, 10 A maximum per feed pair. If the DC power system for the equipment rack is installed with more than 10 amperes of protection, supplemental protection for each server must be provided. The overall current rating of a maximum configured server chassis is 8 amperes.

12.3.2.2 AC Power

Mains AC Power Disconnect

The AC power cords are considered the mains disconnect for each server and must be readily accessible when installed. If the individual server power cords will be be readily accessible for disconnection, an AC power disconnect for the entire rack must be provided. This main disconnect must be readily accessible, and it must be labeled as controlling power to the entire rack, not just to the servers.

Grounding the Rack Installation

To avoid the potential for an electrical shock hazard, a third wire safety ground conductor must be included with the rack installation. If the server power cords are plugged into an AC outlet that is part of the rack, proper grounding must be provided for the rack itself. If the server power cord is plugged into a wall AC outlet, the safety ground conductor in the power cord provides proper grounding only for the server. Additional, proper grounding must be provided for the rack and other devices installed in it. This system is intended for connection to a Common Bonding Network (CBN) as defined by NEBS GR-1089.

Overcurrent Protection

The server is designed to an AC line voltage source with up to 20 amperes of overcurrent protection per cord feed. If the power system for the equipment rack is installed on a branch circuit with more than 20 amperes of protection, supplemental protection must be provided for each server. The overall current rating of a configured server is less than 6 amperes.

Do not attempt to modify or use an AC power cordset that is not the exact type required. The power cordset must meet the following criteria:

- Rating:– In the US and Canada, cords must be UL (Underwriters Laboratories, Inc.) Listed/CSA (Canadian Standards Organization) Certified type SJT, 18-3 AWG (American Wire Gauge). Outside of the US and Canada, cords must be flexible harmonized (<HAR>) or VDE (Verband Deutscher Elektrotechniker, German Institute of Electircal Engineers) certified cord with 3 x 0.75 mm conductors rated 250 VAC.
- Connector, wall outlet end:– Cords must be terminated in grounding-type male plug designed for use in your region. The conenct must have certification marks showing certification by an agency acceptable in your region and for U.S. must be Listed and rated 125% of overall current rating of each server.
- Connector, server end:– The connectors that plug into the AC receptacle on the server must be an approved IEC (International Electrotechnical Commission) 320, sheet C13, type female connector.
- Cord length and flexibility:– Cores must be less than 4.5 meters (14.8 feet) long.

12.3.3 Cooling

Temperature

The temperature, in which each server operates when installed in an equipment rack must not go below 5 degrees Celcius (41 degress Farenheit) or rise above 35 degress Celcius (95 degress Farenheit). Extreme fluctuations in temperature can cause a variety of problems.

Ventilation

The equipment rack must provide sufficient airflow to the front of each server to maintain proper cooling. The rack must also include ventilation sufficient to exhaust a maximum of 1023 BTUs (British Thermal Units) per hour for the server. The rack selected and the ventilation provided must be suitable to the environment in which the servers will be used.

12.4 Spares

Spares will not be provided for the initial trial. Spares will be provisioned for the full in-service date. Spares consist of the following:

- spare Intel Xeon 5400 series processor;
- spare memory modules;
- spare pluggable 450 Watt -48 VDC power supplies;
- spare pluggable fan modules (CPU fans, IO fans);
- spare filter packs;
- spare pluggable SAS harddrives;
- spare *Optranex* 248 SONET/SDH interface PCI Express cards; and,
- spare pluggable OC-12 SFP modules.

Note that with the exception of SONET optical modules, SAS harddrives and power supplies, replacement of parts including processors, memory modules, fan modules, filter packs, and PCI cards requires powering down at least one of the redundant servers. SONET optical modules, SAS harddrives and power supplies are hot-pluggable.

12.5 Schedule

12.5.1 Equipment Delivery and Installation

Optranex server chassis and equipment rails will be delivered to site and installed first. According to current schedules, servers will be shipped to arrive between March 1 and March 30, 2010.

Optranex interface cards, SONET pluggable modules, SMA patch cables, and SFP loop-back cables, will be shipped separately. SONET/SDH Interface cards and components will be shipped, according to current schedules, to arrive between March 30, 2010 and April 12, 2010.

12.5.2 Field Trial

Test plan execution and trunk turn up will commence, according to current schedules, the week of April 12, 2010. During the period from April 12, 2010 to May 12, 2010, the *Optranex* platform may be turned up and down and may experience unplanned service outages. All planned outages will be performed within an preplanned evening maintenance window after trunks have been turned up the week of April 12, 2010. The *Optranex* platform can only be provisioned by *Optranex* staff during the field trial.

12.5.3 Provisioning Trial

Provisioning of the services provided by the *Optranex* 248 will be available, according to current schedules, the week of May 10, 2010. The provisioning plan will be executed that week to prove the integration with back-end provisioning systems. After the provisioning trial is complete, the *Optranex* 248 platform will be provisionable by *Heranax* personnel.

12.5.4 Inservice

In service will commence June 1, 2010. A separate inservice plan will be prepared to address redundancy, spares, etc.

13 Provisioning Plan

13.1 SONET Segment Assignments

13.2 SONET Path Assignments

13.3 SIGTRAN Signaling Gateway Additions

This section provides an overview of the addition of signalling gateways for SIGTRAN connectivity from the Optranex switch to the SS7 signaling network.

13.4 PSTN Signaling Point Additions

This section provides an overview of the addition of remote signaling points for SS7 Signaling over SIGTRAN M3UA.

To assign a signaling point, the following information is required:

1. Signaling point code.
2. Protocol variant.
3. Protocol options. Protocol options include the following:
 - SS7_POPT_UPT — whether to send UPT messages.
 - SS7_POPT_LPA — whether to send LPA messages.
 - SS7_POPT_UCIC — whether to send UCIC messages.
4. Exchange type. This can be one of the following:
 - ISUP_XCHG_TYPE_A — the exchange is a end-office exchange.
 - ISUP_XCHG_TYPE_B — the exchange is a tandem exchange.

13.5 PSTN Signaling Relation Additions

This section provides an overview of the addition of signalling relations for SS7 ISUP Signaling over SIGTRAN M3UA.

Assignments that need to be made prior to signaling relation additions are as follows:

1. Signaling point assignment.

Assignments made to add signaling relations are as follows:

1. Signaling relation assignment.

To assign as signaling relation, the following information is required:

1. The signaling point code of the near-end signaling point. This signaling point is identified by its signaling point identifier (which is an identifier local to the Optranex switch).
2. The signaling point code of the far-end signaling point. This signaling point is identified by its signaling point code.
3. Protocol variant used in the communication between signaling points.
4. Protocol options used between signaling points.
5. Timer values as follows (when they deviate from default values):
 - T4 — waiting for UPA or other messages.

- T18 — waiting for circuit-group-blocking-acknowledge retry.
- T19 — waiting for circuit-group-blocking-acknowledgement maintenance.
- T20 — waiting for circuit-group-unblocking-acknowledge retry.
- T21 — waiting for circuit-group-unblocking-acknowledge maintenance.
- T22 — waiting group-reset-acknowledge retry.
- T23 — waiting group-reset-acknowledge maintenance.
- T28 — waiting circuit-query-response.
- T29 — congestion attack timer.
- T30 — congestion decay timer.
- Tcgb — waiting for blocking-acknowledge (ANSI).
- Tgrs — waiting for reset-acknowledge (ANSI).
- Thga — waiting for blocking-acknowledge hardware (ANSI).
- Tscga — waiting for blocking-acknowledge software (ANSI).
- Tscga_d — waiting for blocking-acknowledge software (ANSI).

13.6 SIP Proxy Additions

This section provides an overview of the addition of Customer proxies to the OpenSIPs and C-SCP platforms. The Optranex is not involved in this assignment.

13.7 PSTN Trunk Group Additions

This section provides an overview of the addition of PSTN trunks to establish trunk group connectivity between the Optranex and remote switches.

Assignments that need to be made prior to PSTN trunk group additions are as follows:

1. SONET segment assignment.
2. SIGTRAN SG assignment.
3. PSTN Signaling Point assignment.

Assignments made to add PSTN trunk groups are as follows:

1. ISUP trunk group assignment.

The attributes (data) necessary to make an ISUP trunk group assignment is as follows:

1. Near-end signaling point code. The near-end signalling point code is a point code selected from the point codes assigned to the *Optranex* switch.
2. Far-end signaling point code. The far-end signalling point code is a point code selected from the point codes assigned to the remote switch to which the trunk group connects.
3. Direction. This can be one of the following:
 - ISUP_TG_TYPE_OG — one-way outgoing trunk group.
 - ISUP_TG_TYPE_IC — one-way incoming trunk group.
 - ISUP_TG_TYPE_2W — both-way trunk group.
4. Circuit selection procedure. This can be one of the following:
 - ISUP_SELECTION_TYPE_MRU — most recently used.
 - ISUP_SELECTION_TYPE_LRU — least recently used.
 - ISUP_SELECTION_TYPE_ASEQ — ascending sequential.

- ISUP_SELECTION_TYPE_DSEQ — descending sequential.
5. Trunk group options. This can be zero or more of the following:
 - ISUP_TGF_GLARE_PRIORITY — when set, indicates that the local exchange has priority in glare conditions; when unset, indicates that the remote exchange has priority in glare conditions.
 - ISUP_TGF_INCOMING_INTERNATIONAL_EXCHANGE — set to indicate that the remote exchange is an incoming international exchange on this trunk group.
 - ISUP_TGF_SUSPEND_NATIONALLY_PERFORMED — set to indicate that suspend and resume procedures are performed according to national specifications.
 - ISUP_TGF_CONTROLLING_EXCHANGE — set to indicate that the local exchange is a controlling exchange on this trunk group.

13.8 PSTN Route Additions

This section provides an overview of the addition of overflow routes. Overflow routes consist of a series of one-way-outgoing or both-way trunk groups. The series identifies the overflow pattern for outgoing circuit seizures. The first trunk group in the series is attempted first. If as circuits are busy in the first group in the series, the second group is attempted.

Assignments that need to be made prior to PSTN route additions are as follows:

1. PSTN trunk group assignment.

Assignments made to add PSTN routes are as follows:

1. PSTN route assignment.

13.9 PSTN Circuit Group Additions

This section provides an overview of the addition, change or removal of circuit groups. Circuit groups are used in the management and maintenance procedures for ISUP signaling. ANSI uses implied or pre-assigned circuit groups. A circuit group is identified using one of the circuit identification codes assigned within the circuit group. Circuit groups must be assigned before circuits within the group can be assigned.

Assignments that need to be made prior to circuit group assignments are as follows:

1. PSTN Signaling Point assignments (both local and remote).

Assignments made to add circuit groups are as follows:

1. PSTN circuit group assignment.

13.10 PSTN Circuit Additions

This section provides an overview of the addition, change or removal of trunks (circuits) within PSTN trunk groups.

Assignments that need to be made prior to PSTN trunk additions are as follows:

1. PSTN trunk group assignment.
2. PSTN circuit group assignment.
3. SONET path assignment.

Assignments made to add PSTN trunk groups are as follows:

1. Near-end signaling point code.

2. Far-end signaling point code.
3. Primary circuit identification code.
4. Pre-assigned or implied grouping.
1. ISUP circuit assignment.

13.11 PSTN Virtual Trunk Group Additions

This section provides an overview of the addition, change or removal of PSTN virtual trunk groups (for outgoing calls only).

13.12 SIP Virtual Trunk Group Additions

This section provides an overview of the addition of SIP virtual trunk groups (for incoming calls only) to the OpenSIPs and C-SCP platforms.

13.12.1 OpenSIPS

13.12.2 Optranex

13.12.3 C-SCP

14 Test Plan

14.1 OpenSIPS

14.2 Optranex

14.3 C-SCP

14.4 Outgoing Calls

14.5 Incoming Calls

Appendix A Alternate Hardware

A.1 Alternate SFP Modules

This appendix contains lists of SFP modules that are compatible with the *Optranex 248* card according to their advertised specifications. Note that these modules have not been tested; however, if they are compliant with the SFP MSA as advertised, they should work properly with the *Optranex 248*.

A.1.1 Eoptolink Modules

Suitable **Eoptolink** SFP modules are listed below in [Table A.1](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
EOLS-1326-02	OC-48/12/3	S	1310	2			
EOLS-1326-X	OC-48/12/3	S	1310	40			
EOLS-1526-X	OC-48/12/3	S	1550	80			
EOLS-1324-02	OC-48	S	1310	2			
EOLS-1324-X	OC-48	S	1310	40			
EOLS-1524-X	OC-48	S	1550	100			
EOLS-1306-X	OC-12	S	1310	40			
EOLS-1506-X	OC-12	S	1550	80			
EOLS-1303-X	OC-3	S	1310	60			
EOLS-15-3-X	OC-3	S	1550	120			

Table A.1: *Eoptolink SFP Modules*

Eoptolink also has a number of OC-3 to OC-48 capable single fiber bidirectional WDM SFP modules, however, these are not in common usage for central office equipment.

A.1.2 Xgiga SFP Modules

Suitable **Xgiga** SFP modules are listed below in [Table A.2](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
XGSF-1524-80D	OC-48	S	1550	80			
XGSF-1524-40D	OC-48	S	1550	40			
XGSF-1324-40D	OC-48	S	1310	40			
XGSF-1324-15D	OC-48	S	1310	15			
XGSF-1503-80	OC-3	S	1550	80			
XGSF-1303-40	OC-3	S	1310	40			
XGSF-1303-15D	OC-3	S	1310	15			
XGSF-1303-02	OC-3	S	1310	2			

Table A.2: *Xgiga SFP Modules*

A.1.3 Star Opto SFP Modules

Suitable **Star Opto** SFP modules are listed below in [Table A.3](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
SSFP3161-13-125+	OC-48	S	1310	2			
SSFP3261-23-125+	OC-48	S	1310	15			
SSFP5261-23-125+	OC-48	S	1550	40			
SSFP5561-33-125+	OC-48	S	1550	80			
SSFP3251-13-125	OC-24	S	1310	30			
SSFP5251-13-125	OC-24	S	1550	60			
SSFP5251-23-125	OC-24	S	1550	80			
SSFP3141-13-125+	OC-12	S	1310	20			
SSFP3141-23-125+	OC-12	S	1310	40			
SSFP5241-13-125+	OC-12	S	1550	40			
SSFP5241-23-125+	OC-12	S	1550	80			
SSFP3131-13-125+	OC-3	S	1310	20			
SSFP3131-23-125+	OC-3	S	1310	40			
SSFP3231-43-125+	OC-3	S	1310	80			
SSFP5131-13-125+	OC-3	S	1550	20			
SSFP5131-43-125	OC-3	S	1550	100			
SSFP5231-13-125+	OC-3	S	1550	40			
SSFP5231-23-125+	OC-3	S	1550	100			

Table A.3: *Star Opto SFP Modules***A.1.4 Opnext SFP Modules**

Suitable **Opnext** SFP modules are listed below in [Table A.4](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
TRF591x	OC-48	S	1310	2			
TRF592x	OC-48	S	1310	15			
TRF595x	OC-48	S	1310	40			
TRF796x	OC-48	S	1550	80			
TRF552x	OC-12	S	1310	15			
TRF555x	OC-12	S	1310	40			
TRF756x	OC-12	S	1550	80			
TRF758x	OC-12	S	1550	100			
TRF542x	OC-3	S	1310	15			
TRF545x	OC-3	S	1310	40			
TRF746x	OC-3	S	1550	80			
TRF748x	OC-3	S	1550	100			

Table A.4: *Opnext SFP Modules***A.1.5 NEC SFP Modules**

Suitable **NEC** SFP modules are listed below in [Table A.5](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
OD-J8836-5C01	OC-48	S	1310	2			
OD-J8837-5C01	OC-48	S	1310	15			
OD-J8838-5C01	OC-48	S	1550	40			
OD-J8839-5C01	OC-48	S	1550	80			

Table A.5: *NEC SFP Modules*

A.1.6 LuminentOIC SFP Modules

Suitable **LuminentOIC** SFP modules are listed below in [Table A.6](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Avnet
SP-MR-SR1	OC-48/12/3	S	1310	2			
SP-MR-IR1	OC-48/12/3	S	1310	15			
SP-MR-IR2	OC-48/12/3	S	1550	40			
SP-MR-LR1	OC-48/12/3	S	1310	40			
SP-MR-LR2	OC-48/12/3	S	1550	80			
SP-DR-IR1	OC-12/3	S	1310	15			
SP-12-IR1	OC-12	S	1310	15			
SP-12-LR1	OC-12	S	1310	40			
SP-12-LR2	OC-12	S	1550	80			
SP-03-IR1	OC-3	S	1310	15			
SP-03-LR1	OC-3	S	1310	40			
SP-03-LR2	OC-3	S	1550	80			

Table A.6: *LuminentOIC SFP Modules*

A.1.7 GEVISTA SFP Modules

Suitable **GEVISTA** SFP modules are listed below in [Table A.7](#).

Part Number	Rate	FM	nm	km	Digikey PN	Price	Newstar
SFP-OC48-SR	OC-48	S	1310	2			\$125.00
SFP-OC48-IR1	OC-48	S	1310	15			\$110.00
SFP-OC48-LR1	OC-48	S	1310	40			
SFP-OC48-LR2	OC-48	S	1550	80			\$330.00
SFP-OC12-MM	OC-12	M	1310	2			\$85.00
SFP-OC12-SR	OC-12	S	1310	2			\$43.00
SFP-OC12-IR1	OC-12	S	1310	15			\$59.00
SFP-OC12-LR1	OC-12	S	1310	40			\$115.00
SFP-OC12-LR2	OC-12	S	1550	80			\$150.00
SFP-OC3-MM	OC-3	M	1310	2			\$39.00
SFP-OC3-SR	OC-3	S	1310	2			\$50.00
SFP-OC3-IR1	OC-3	S	1310	15			\$70.00
SFP-OC3-LR1	OC-3	S	1310	40			\$103.00
SFP-OC3-LR2	OC-3	S	1550	80			\$125.00

Table A.7: *GEVISTA SFP Modules*

These modules are available from [Newstar](#).

Appendix B Protocol Values

B.1 Cause Values

This section describes cause values applicable to US networks and how they interact with SIP 4xx, 5xx and 6xx responses. In these responses, the cause value is contained in the **Reason** header in accordance with *RFC 3326*.¹

B.1.1 Cause Value to SIP Response Mapping

When the *Optranex 248* gateway switch generates a SIP 4xx, 5xx or 6xx response to an **INVITE** message from the SIP network, no **Reason** header will be included in the SIP response. This is to avoid the situation where an attack on the platform is assisted by the detailed cause value in a response message.

When the *Optranex 248* gateway switch receives a SIP 4xx, 5xx, or 6xx response to an **INVITE** message sent to the SIP network, the **Reason** header and the contained cause values will only be honored if the response was sent via the *C-SCP*. That is, a *C-SCP* host name or IP address is contained in a **Via** header in the response immediately after the *Carrier Proxy Via* header. This is to avoid attacks on the platform by UAs that arbitrarily set a *Reason* header in a response.

- ITU-T Q.850 Cause 1 *Unallocated (unassigned) number*. This cause indicates that the called party cannot be reached because, although the called party number is in a valid format, it is not currently allocated (assigned).

This cause value can be contained in (and will trigger) a **404 Not Found** response. This is the default value that will be assigned if a **404 Not Found** message is received without an **ANSI** or **Q.850 Reason** header. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 2 *No route to specified transit network*. This cause indicates that the equipment sending this cause has received a request to route the call through a particular transit network which it does not recognize. The equipment sending this cause does not recognize the transit network either because the transit network does not exist or because that particular transit network, while it does exist, does not serve the equipment which is sending this cause. This cause is supported on a network-dependent basis.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 3 *No route to destination*. This cause indicates that the called party cannot be reached because the network through which the call has been routed does not serve the destination desired. This cause is supported on a network-dependent basis.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 4 *Send special information tone*. This cause indicates that the calling party cannot be reached for reasons that are of a long term nature and that the special information tone should be returned to the calling party.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

¹ One extension to *RFC 3326* is provided in that the **ANSI** token will be recognized in addition to the **Q.850** token.

- ITU-T Q.850 Cause 5 *Misdialled trunk prefix*. **This cause value is not used for US networks.**
This cause value can be contained in (and will trigger) a 404 **Not Found** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 8 *Preemption*. This cause indicates that the call is being preempted.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 9 *Preemption - circuit reserved for reuse*. This cause indicates that the call is being preempted and the circuit is reserved for reuse by the preempting exchange.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 16 *Normal call clearing*. This cause is used to indicate that the call is being cleared because one of the users involved in the call has requested that the call be cleared. Under normal situations, the source of this cause is not the network.
This cause value is indicated by (or triggers) a **BYE** or **CANCEL** request. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 17 *User busy*. This cause is used to indicate that the called party is unable to accept another call because the user busy condition has been encountered. This cause value may be generated by the called user or by the network. In the case of user determined user busy, it is noted that the user equipment is compatible with the call.
This cause value is used whenever a 486 **Busy Here** response is received. This cause value can be contained in (and will trigger) a 486 **Busy Here** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 18 *No user responding*. This cause is used when a called party does not response to a call establishment message with either an alerting or connect indication within the prescribed period of time allocated.
This cause value can be contained in (and will trigger) a 480 **Temporarily Unavailable** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 19 *No answer from user (user alerted)*. This value is used when the called party has been alerted but does not response with a connect indication within the prescribed period of time. Note - this cause is not necessarily generated by Q.931 procedures but may be generated by internal network timers.
This cause value can be contained in (and will trigger) a 480 **Temporarily Unavailable** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 20 *Subscriber absent*. This cause value is used when a mobile station has logged off, radio contact is not obtained with a mobile station or if a personal telecommunication user is temporarily not addressable by any user-network interface.
This cause value can be contained in (and will trigger) a 480 **Temporarily Unavailable** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 21 *Call rejected*. This cause indicates that the equipment sending this cause does not wish to accept this call, although it could have been accepted the call because the equipment sending this cause is neither busy nor incompatible. This cause may also be generated by the network, indicating that the call was cleared due to a supplementary service constraint. The diagnostic field may contain additional information about the supplementary service and reason for rejection.
This cause value can be contained in (and will trigger) a 480 **Temporarily Unavailable** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 22 *Number changed*. This cause is returned to a calling party when the called party number indicated by the calling party is no longer assigned. The new called party number may optionally be included in the diagnostic field. If a network does not support this cause value, cause number 1, unallocated (unassigned) number shall be used.

This cause value is used whenever a **410 Gone** response is received. This cause value is used whenever a **301 Moved Permanently** response is received. The **tel** URI contained in the **Contact** header contains the new number. This cause value can be contained in (and will trigger) a **410 Gone** or **301 Moved Permanently** response (depending upon whether the number is present). The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 23 *Redirect to new destination*. This cause is used by a general ISUP protocol mechanism that can be invoked by an exchange that decides that the call should be set up to a different called number. Such an exchange can invoke a redirect mechanism, but use of this cause value, to request a preceding exchange involved in the call to route the call to the new number.

This cause value is used whenever a **302 Moved Temporarily** response is received. The **tel** URI contained in the **Contact** header contains the new number. This cause value can be contained in (and will trigger) a **302 Moved Temporarily** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 27 *Destination out of order*. This cause indicates that the destination indicated by the user can not be reached because the interface to the destination is not functioning correctly. The term "not functioning correctly" indicates that a signaling message was unable to be delivered to the remote party. For example, a physical layer or data link layer failure at the remote party, or user equipment off-line.

This cause value is used whenever a **502 Bad Gateway** response is received. This cause value can be contained in (and will trigger) a **502 Bad Gateway** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 28 *Invalid number format (address incomplete)*. This cause indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete. Note - this condition may be determined immediately after the reception of an ST signal, or on time-out after the last received digit.

This cause value is used whenever a **484 Address Incomplete** response is received. This cause value can be contained in (and will trigger) a **484 Address Incomplete** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 29 *Facility rejected*. This cause is returned when a supplementary service requested by the user cannot be provided by the network.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 31 *Normal, unspecified*. This cause is used to report a normal event only when no other cause in the normal class applies.

This cause value can be contained in (and will trigger) a **480 Temporarily Unavailable** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 34 *No circuit/channel available*. This cause indicates that there is no appropriate circuit/channel presently available to handle the call.

This cause value can be contained in (and will trigger) a **480 Temporarily Unavailable** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.

- ITU-T Q.850 Cause 38 *Network out of order*. This cause indicates that the network is not functioning correctly and that the condition is likely to last a relatively long period of time; for example, immediately re-attempting the call is not likely to be successful.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 41 *Temporary failure*. This cause indicates that the network is not functioning correctly and that the condition is not likely to last a long period of time; for example, the user may wish to try another call attempt almost immediately.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 42 *Switching equipment congestion*. This cause indicates that the switching equipment generating this cause is experiencing a period of high traffic.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 43 *Access information discarded*. This cause indicates that the network could not deliver access information to the remote user as requested; that is, user-to-user information, low layer compatibility, high layer compatibility, or sub-address, as indicated in the diagnostic.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 44 *Requested circuit/channel not available*. This cause is returned when the circuit or channel indicated by the requesting entity cannot be provided by the other side of the interface.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 46 *Precedence call blocked*. This cause indicates that there are no preemptable circuits or that the called user is busy with a call of equal or higher preemptable level.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 47 *Resource unavailable, unspecified*. This cause is used to report a resource unavailable event only when no other cause in the resource unavailable class applies.
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is expected to generate this cause value under appropriate conditions.
- ITU-T Q.850 Cause 50 *Requested facility not subscribed*. This cause indicates that the user has requested a supplementary service which is implemented by the equipment which generated the cause, but the user is not authorized to use.
This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 53 *Outgoing calls barred within CUG*. **This cause value is not used in US networks.**
This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 55 *Incoming calls barred within CUG*. **This cause value is not used in US networks.**

This cause value can be contained in (and will trigger) a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 57 *Bearer capability not authorized*. This cause indicates that the user has requested a bearer capability which is implemented by the equipment which generated this cause but the user is not authorized to use.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 58 *Bearer capability not presently available*. This cause indicates that the user has requested a bearer capability which is implemented by the equipment which generated this cause but which is not available at this time.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 62 *Inconsistency in designated outgoing access information and subscriber class*. This cause indicates that there is an inconsistency in the designated outgoing access information and subscriber class.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 63 *Service or option not available, unspecified*. This cause is used to report a service or option not available event only when no other cause in the service or option not available class applies.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 65 *Bearer capability not implemented*. This cause indicates that the equipment sending this cause does not support the bearer capability requested.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 69 *Requested facility not implemented*. This cause indicates that the equipment sending this cause does not support the requested supplementary service.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 70 *Only restricted digital information bearer capability is available*. This cause indicates that the calling party has requested an unrestricted bearer service but that the equipment sending this cause only supports the restricted version of the requested bearer capability.

This cause value can be contained in a 500 **Server Internal Error**, 606 **Not Acceptable** or 488 **Not Acceptable** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 79 *Service or option not implemented, unspecified*. This cause is used to report a service or option not implemented event only when no other cause in the service or option not implemented class applies.

This cause value can be contained in a **500 Server Internal Error**, **606 Not Acceptable** or **488 Not Acceptable** response, and will trigger a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 87 *User not member of CUG*. **This cause value is not used in US networks.**
This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 88 *Incompatible destination*. This cause indicates that the equipment sending this cause has received a request to establish a call which has low layer compatibility, high layer compatibility, or other attributes (e.g. data rate) which cannot be accommodated.
This cause value can be contained in a **500 Server Internal Error**, **606 Not Acceptable**, or **488 Not Acceptable** response, and will trigger a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 90 *Non-existent CUG*. **This cause value is not used in US networks.**
This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 91 *Invalid transit network selection*. This cause value indicates that a transit network identification was received which is of an incorrect format as defined in Annex C of T1.607.
This cause value can be contained in (and will trigger) a **404 Not Found** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 95 *Invalid message, unspecified*. This cause is used to report an invalid message event only when no other cause in the invalid message class applies.
This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 97 *Message type non-existent or not implemented*. This cause indicates that the equipment sending this cause has received a message with a message type it does not recognize either because this is a message not defined or defined but not implemented by the equipment sending this cause.
This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 99 *Information element/Parameter non-existent or not implemented*. This cause indicates that the equipment sending this cause has received a message which includes information elements or parameters not recognized because the information element identifiers or parameter names are not defined or are defined but not implemented by the equipment sending the cause. This cause indicates that the information elements or parameters were discarded. However, the information element is not required to be present in the message for the equipment sending the cause to process the message.
This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 102 *Recovery on timer expiry*. This cause indicates that a procedure has been initiated by the expiry of a timer in association with error handling procedures.
This cause value can be contained in (and will trigger) a **480 Temporarily Unavailable** response. The *C-SCP* is not expected to generate this cause value.
- ITU-T Q.850 Cause 103 *Parameter non-existent or not implemented - passed on*. This cause indicates that the equipment sending this cause has received a message that includes parameter

not recognized because the parameters are defined but not implemented by the equipment sending the cause. The cause indicates that the parameters were ignored. In addition, if the equipment sending this cause is an intermediate point, then this cause indicates that the parameters were passed on unchanged.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 110 *Message with unrecognized parameter discarded*. This cause indicates that the equipment sending this cause has discarded a received message that includes a parameter that is not recognized.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 111 *Protocol error, unspecified*. This cause is used to report a protocol error event only when no other cause in the protocol error class applies.

This cause value can be contained in (and will trigger) a **500 Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ITU-T Q.850 Cause 127 *Interworking, unspecified*. This cause indicates that there has been interworking with a network that does not provide causes for actions it takes. Thus, the precise cause for a message which is being sent cannot be ascertained. This cause value will be sent to the ISUP network whenever a **4xx** or **5xx** response is received with no **Reason** header containing an ANSI or Q.850 header value, and the **4xx** or **5xx** message does not directly map to an ANSI or Q.850 cause value.

This cause value can be contained in (and will trigger) a **480 Temporarily Unavailable** response. The *C-SCP* is not expected to generate this cause value.

- ANSI T1.113/2000 Cause 23 *Unallocated destination number*. This cause indicates failure of a business group call because the destination number is unallocated in the business group numbering plan.

This cause value can be contained in (and will trigger) a **404 Not Found** response. The *C-SCP* is not expected to generate this cause value.

- ANSI T1.113/2000 Cause 24 *Unknown business group*. This cause indicates failure of a business group call because the business group identifier is unknown.

This cause value can be contained in (and will trigger) a **404 Not Found** response. The *C-SCP* is not expected to generate this cause value.

- ANSI T1.113/2000 Cause 25 *Exchange routing error*. This cause indicates that the called party cannot be reached because of an error in exchange number translation tables used to route the call (results when the hop counter parameter value is decremented to 0).

This cause value will be used whenever a **483 Too Many Hops** response or a **482 Loop Detected** response is received. This cause value can be contained in a **483 Too Many Hops**, **482 Loop Detected**, or **480 Temporarily Unavailable** response, and will trigger a **480 Temporarily Unavailable** response.

- ANSI T1.113/2000 Cause 26 *Misrouted call to a ported number*. This cause indicates that the called party cannot be reached because the "ported number" in the generic address parameter identifies a called party that is not served by the exchange.

This cause value can be contained in a **500 Server Internal Error** response or a **404 Not Found** response, and will trigger a **500 Server Internal Error** response. The *C-SCP* must generate this cause value under appropriate conditions.

- ANSI T1.113/2000 Cause 27 *Number portability Query on Release (QoR) number not found.* **This cause value is not used in US networks.**

This cause value can be contained in a 500 **Server Internal Error** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* must generate this cause value under appropriate conditions.

- ANSI T1.113/2000 Cause 45 *Preemption.* **This cause value is not used in US networks.** This cause indicates that the equipment sending this cause has preempted the circuit for a new call and the existing call should be cleared.

This cause value can be contained in a 500 **Server Internal Error** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ANSI T1.113/2000 Cause 46 *Precedence call blocked.* **This cause value is not used in US networks.** In case of a call with a precedence level higher than the lowest level (ROUTINE), this cause indicates that all circuits are busy on calls of equal or higher priority.

This cause value can be contained in a 500 **Server Internal Error** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ANSI T1.113/2000 Cause 51 *Call type incompatible with service request.* This cause indicates that an action was rejected because it was not compatible with the bearer capability of the call. This cause value can be contained in a 500 **Server Internal Error** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

- ANSI T1.113/2000 Cause 54 *Call blocked due to group restrictions.* This cause indicates failure of a business group call because it did not pass line privileges and/or subgroup screening.

This cause value can be contained in a 500 **Server Internal Error** response, and will trigger a 500 **Server Internal Error** response. The *C-SCP* is not expected to generate this cause value.

B.1.2 Default SIP Response to Cause Value Mapping

When the *Optranex 248* gateway switch receives a SIP 4xx, 5xx, or 6xx response to an INVITE message sent to the SIP network, and the *Via* headers do not indicate that the response was generated by the *C-SCP* platform, the SIP response will be mapped to a default cause value associated with the response type.

- **301 Moved Permanently** The cause value ITU-T Q.850 Cause 22 *Number changed* will be assigned. It is not expected that the *Optranex 248* switch will receive 3xx class messages. 3xx class messages should be intercepted by the *Carrier Proxy*.
- **302 Moved Temporarily** The cause value ITU-T Q.850 Cause 23 *Redirect to new destination* will be assigned. Should the **Contact** header not be decipherable as a valid **tel** URI, the cause value ITU-T Q.850 Cause 1 *Unallocated (unassigned) number* will be assigned instead. It is not expected that the *Optranex 248* switch will receive 3xx class messages. 3xx class messages should be intercepted by the *Carrier Proxy*.
- **3xx** The cause value ITU-T Q.850 Cause 127 *Interworking, unspecified* will be assigned when an unrecognized 3xx message or a 3xx message for which no default is assigned is received. It is not expected that the *Optranex 248* switch will receive 3xx class messages. 3xx class messages should be intercepted by the *Carrier Proxy*.
- **404 Not Found** The cause value ITU-T Q.850 Cause 1 *Unallocated (unassigned) number* will be assigned.

- **410 Gone** The cause value ITU-T Q.850 Cause 22 *Number Changed* will be assigned.
- **480 Temporarily Unavailable** The cause value ITU-T Q.850 Cause 20 *Subscriber absent* or ITU-T Q.850 Cause 21 *Call rejected* will be assigned.
- **482 Loop Detected** The cause value ITU-T Q.850 Cause 25 *Exchange routing error* will be assigned.
- **483 Too Many Hops** The cause value ITU-T Q.850 Cause 25 *Exchange routing error* will be assigned.
- **484 Address Incomplete** The cause value ITU-T Q.850 Cause 28 *Invalid number format (address incomplete)* will be assigned.
- **486 Busy Here** The cause value ITU-T Q.850 Cause 17 *User busy* will be assigned.
- **488 Not Acceptable Here** The cause value ITU-T Q.850 Cause 63 *Service or option not available, unspecified* will be assigned.
- **4xx** The cause value ITU-T Q.850 Cause 127 *Interworking, unspecified* will be assigned when an unrecognized 4xx message or a 4xx message for which no default is assigned is received.
- **500 Server Internal Error** The cause value ITU-T Q.850 Cause 38 *Network out of order* will be assigned.
- **502 Bad Gateway** The cause value ITU-T Q.850 Cause 27 *Destination out of order* will be assigned.
- **503 Service Unavailable** The cause value ITU-T Q.850 Cause 41 *Temporary failure* will be assigned.
- **5xx** The cause value ITU-T Q.850 Cause 127 *Interworking, unspecified* will be assigned when an unrecognized 5xx message or a 5xx message for which no default is assigned is received.
- **603 Decline** The cause value ITU-T Q.850 Cause 21 *Call rejected* will be assigned.
- **604 Does Not Exist Anywhere** The cause value ITU-T Q.850 Cause 1 *Unassigned (unallocated) number* will be assigned.
- **606 Not Acceptable** The cause value ITU-T Q.850 Cause 63 *Service or option not available, unspecified* will be assigned.
- **6xx** The cause value ITU-T Q.850 Cause 127 *Interworking, unspecified* will be assigned when an unrecognized 6xx message or a 6xx message for which no default is assigned is received.

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Appendix D Platform Sizing

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